*“**A**s a computer scientist and musician, I’m enjoying designing DSP algorithms and plug-in concepts. It’s a pleasure for me to share my efforts with other people without having to worry about commercial interests. In each of my effects there is a special technical design, workflow or concept combination I haven’t found in existing tools. Many developers these days just stick to old concepts, but you will never see a straight copy of existing gear from me. I always add some innovation, or find a way to evolve and take it to a new level.”*

Variety of Sound

sound design & audio effect programming

BootEQ mkII

Usage tips:

● Use the 'OUT' knob to level the outgoing audio and for handy A/B comparisons
(works just for the pre-amp section, not for the EQ)
● Use <ctrl> + mouse left click on a knob or switch to restore default position
● Use <shift> + mouse left click on a knob to fine adjust values
● Use this plugin as an insert effect in any mono or stereo channel of your VST host
● Use the presets just as a basic reference: EQing is to be an individual approach each time, there is no magic setting

Some general tips on EQing (related to mixing, not mastering):

● Use your ears and not your eyes. You will make different EQ decisions either done by eye or by ear, but the hearing rulez
● Sweep through frequency spectrum with high EQ gain/peak settings to identify resonant or unpleasant frequencies, but:
● Make your specific EQ decision always in context of the rest of the mix. EQing is always relative and not absolute
● Use coloring EQ's to your advantage to obtain certain sound qualities whilst mixing (if necessary and wanted)
● Use technical EQ's for steep and surgical corrections

And always remember: garbage in, garbage out ;-)

On CPU usage

All different selectable EQ bands are increasing slightly the overall CPU consumption of the plug-in. The other way around disabling them will save CPU cycles. There is just a slight overall overhead and you can disable the whole EQ section if not needed.

Activating the pre-amp section consumes higher CPU usage due to the complexity of the algorithms and the 4x oversampling used here.

Overview

'BootEQ' – a musical sounding mixing EQ and pre-amp simulation.

At a glance:
● four parametric and independent EQ bands
● special selected and musical sounding EQ curves and phase responses
● capable of reproducing several 'classic' curves and EQ behavior
● well adjusted auto Q and versatile overlapping frequency ranges
● minimized curve warping near Nyquist frequency
● detailed modeled pre-amp simulation
● subtle and nice audio coloration enhancements

Plug-in specification:

● PC / VST compatible
● SSE and Assembler optimized sound engine
● state-of-the-art digital signal processing
● low CPU EQ and minimum latency processing
● musical sounding EQ curves (frequency and phase response)
● signal modeled pre-amp simulation

Quick reference

The EQ.

Q: Alter the HF frequency response (slope shape): Turning counter-clockwise broadens towards mid freq's. Turning clockwise features more 'air'. This affects the High Frequency shelving filter: Boost or attenuate frequencies around 10kHz up or down to
12dB.

High Middle Frequency EQ (bell shape): Boost or attenuate frequencies up or down to 12dB. Select the center frequency step-less from 800Hz up to 8.9 kHz.

Low Middle Frequency EQ (bell): Boost or attenuate frequencies up or down to 12dB. Select the center frequency step-less from 100Hz up to 1.5kHz. There is a switch on the left to change from steeper to broader curves.

Low Frequency filter section: This peak filter is switchable to high pass mode. The peak filter allows -12 to +12dB adjustments from 40 to 250Hz. In high pass mode the gain dial changes the curve and steepness of the filter.

ON/OFF turns the whole EQ on or off.

The pre-amp simulator

The VU style meter displays the internal gain level after the DRV dial. Hitting slightly the red metering area should be 'safe' distortion-wise. Hitting constantly the red mark causes audible distortion (all depending on the source material).

The DRV (drive) knob sets the internal and volume compensated gain level of the pre-amp model and changes the overall saturation behavior.
This introduces subtle or audible distortion depending on input level and source.

TUBE ON/OFF: Adds/removes tube style 2nd order harmonics.

VINTAGE/MODERN: Changes the frequency and phase response of the simulation as well as the HF saturation behavior.

LF: Low frequency transformer simulation response. Alters frequency and phase response plus the harmonic audio structure as well.

OUT: boosts or attenuates the outgoing level up or down to 12dB.

ON/OFF turns the whole pre-amp on or off.

Nasty Signal Coloring FX Suite

Usage tips:

- *Use the 'OUT' knob to level the outgoing audio and for handy A/B comparisions*
- *Use <ctrl> + mouse left click on a knob or switch to restore default position*
- *Use <shift> + mouse left click on a knob to fine adjust values*

Some general tips on EQ'ing (related to mixing, not mastering):

1. Use your ears and not your eyes! You will make different EQ decisions either done by eye or by ear. But the hearing rulez.

2. Sweep through frequency spectrum with higher EQ boosts to identify more easily weak or hot spots – but:

3. Always EQ in context!
Make your specific EQ decision always in context of the rest of the mix. EQing is always relative, never absolute.

4. Use coloring devices to your advantage to obtain certain sound qualities whilest mixing (if necessary and wanted).

And always remember: garbage in, garbage out ;-)

Nasty LF

Overview

'NastyLF' - adding mojo to the lowend.

At a glance:

Getting the lowend right is one important key in successfully mixing modern music these days. As a creative mixing device 'NastyLF' offers subtle low frequency enhancements including creamy lowend distortion up to more agressive filtering and nasty saturation.

Concept

'NastyLF' is a specific and tuned combination of lowend EQ and output stage.
The low frequency EQ in classic boost/cut design offers both: broad 'oldschool' as well as rather narrow 'modern' curves (switchable) altogether with a variation of that special sounding curve designs when using boost and cut in combination. A new developed output stage offers tasty lowend saturation which can virtually be driven up to 24dB with internal automatic gain compensation.

Example Applications:

- changing easily tonal balance of lower frequency audio content in a musical sounding fashion
- improving bass presence due to saturation effects
- getting the lowend perception more solid and homogeneous
- adding sonic grip and fatness to thin sounding audio sources

Tech notes:

- offering even and odd harmonics
- zero latency processing
- low CPU usage
- plug-in integration is done with Synthmaker software
- performance crucial parts are written in assembler or optimized by hand
- completely SSE optimized

General usage tips

Use this Plug-In as an insert effect in any mono or stereo channel of your VST host. Assure the 'IN' switch is in upper position (so LED is lightning red). If it's grey drag that switch into upper position. This toggles the overall operation (on/off).

Tip: *Use the 'OUT' knob to adjust the overall output level to equal levels and then use the 'IN' switch for convenient A/B testing (by switching from 'IN' positon to '0' and back).*

How to start?

Use the presets just for some basic orientation or understanding: EQing is to be an individual approach each time, there is no magic setting which fixes general problems. Therefore some more hints here on how to start with this device and how to get the most out of it:

Decide first wether your are more after frequency balancing/correction or more after saturation. If both, start first with frequency adjustments and then apply saturation effects afterwards.
When boosting, always apply some (smaller) amount of cuts to obtain that EQ curve 'dip' and it's musical qualities.

If you are in need of some 'steeper' cutting action this is the wrong device -> choose a technical EQ instead or apply one afterwards.
It's always a good practise to start with a more musical/coloring device followed (if necessary) by more technical (transparent and precise) devices.

General usage tips

Frequency and narrow/wide curves: If one is after a more 'vintage' type of sound I would recommend to start with wide curves (W position). To obtain more modern characteristics choose always the narrow (N) option. Sweep through the frequency afterwards to find spots where the effects are most helpful/pleasant.

Start saturation always in '0' position which adds just minimum effects. If that still is too much (e.g. on sensitive acoustic recordings) you can lower the output (which 'drives' the output stage too) or lower the input signal in your host or maybe you should choose a more subtle working Plug-in (e.g. TesslaSE).

Increase amounts of saturation as needed/wanted. This can also be used for 'maxi-mizing' purposes if some distortion is acceptable. Use the 'LF' switch position to addi-tionally and critically judge the applied distortion just in the context of low and low-mid frequencies.

Tip: *Use the 'IN' switch's lower 'LF' position to just hear the processed signals lower and lower-mid content. This is usefull to better judge a certain impact of settings on the lower frequency content (especially distortion).*

Quick reference

This meter displays the out-going signals level.

Turning the 'BOOST' knob clock-wise increases LF audio content.

Turning the 'CUT' knob clock-wise decreases LF audio content.

Volume control of the output stage. Center position is 0 dB.

'FREQ' selects the center frequency of both 'CUT' and 'BOOST'

Turns the processing 'IN' (LED is red lightning) or out (LED is grey and switch in '0' position). 'LF' position monitors the (processed) LF signals only (LED is flashing then).
Controls the saturation of the output stage in 5 gain steps. Output levels are internally compensated. Position 'OFF' disables the ouput stage effect.

The 'N-W' switch selects more narrow or wide EQ curve behaviour.

Nasty HF

Overview

'NastyHF' - adding mojo to the highend.

At a glance:

Getting the highend right is one important key in successfully mixing modern music these days. As a creative mixing device 'NastyHF' offers pristine high frequency improvements including fancy harmonic enhancements with extrem low artifacts.

Concept

'NastyHF' is a specific and tuned combination of highend EQ (peak and shelve) and output stage. The high frequency EQ offers both: broad 'old school' as well as rather narrow 'mod- ern' curves (switchable). HF peaking is applied on fixed frequencies and the 10kHz filter performs as a shelve. Both designs feature special musical sounding curves.
A new developed output stage offers tasty HF saturation featuring prominently fancy K2 and K3 harmonics with attenuated higher harmonics.

Example Applications

- improving easily the brilliance of recordings and mixes
- improving presence perception due to saturation effects
- getting the high end more “in the face”

- adding sonic grip and warmth to thin sounding audio sources

Tech notes

- oversampled output stage
- featuring K2 and K3 harmonics almost artifact free
- minimum latency processing
- reasonable CPU usage
- Plug-in integration is done with Synthmaker software
- performance crucial parts are written in assembler or optimized by hand
- completely SSE optimized

General usage tips

Use this Plug-In as an insert effect in any mono or stereo channel of your VST host. Assure the 'IN' switch is in upper position (so LED is lightning red). If it's grey drag that switch into upper position. This toggles the overall operation (on/off).

Tip: *Use the 'OUT' knob to adjust the overall output level to equal levels and then use the 'OFF-ON' switch for convenient A/B testing (by switching from 'OFF' positon to 'ON' and back).*

NastyHF ain't that complicated and offers just few controls. However:

When mixing be clear about your current task - analyse and decide which frequency range would be most helpful to boost in your current mix.

If you are in need of some 'steeper' cutting action this is the wrong device -> choose a technical EQ instead or apply one afterwards.
It's always a good practise to start with a more musical/coloring device followed (if necessary) by more technical (transparent and precise) devices.

Frequency and narrow/wide curves: If one is after a more 'vintage' type of sound I would recommend to start with wide curves (W position). To obtain more modern characteristics choose always the narrow (N) option. Sweep through the frequency afterwards to find spots where the effects are most helpful/pleasant.

Quick reference

This meter displays the out- going signals level.

Turning the 'BOOST' knob clock-wise increases HF audio content.

'FREQ' selects the center frequency of the filter. Up to 5kHz is peaking and above is shelving characteris- tics.

Volume control of the output stage. Center position is 0 dB.

The 'N-W' switch selects more narrow or wide EQ curve behavior.

Turns the processing on or off.

NASTY table Top

Overview

'NASTYtableTop' – signal coloring made easy.

At a glance:

Digital audio sources often lacking 'phatness' and impact these days.
As a creative mixing device 'NASTYtableTop' offers easy to use subtle to drastically low-mid frequency boosts.

Concept

'NASTYtableTop' is a low-mid frequency booster which increases perception in this frequency range by saturation. In subtle amounts this is perceived as a kind of 'phat-ness' while in extrem settings this is going to be perceived as 'muddy' and/or 'dis-torted'.

Example Applications

- pimp lame softsynths
- improving bass presence due to saturation effects
- improving overall perception of lower-mid frequencies
- adding sonic grip and fatness to thin sounding audio recordings

Tech notes

- zero phase and zero latency processing
- fool proof single knob design
- low CPU usage
- Plug-in integration is done with Synthmaker software
- performance crucial parts are written in assembler or optimized by hand
- completely SSE optimized

General usage tips

Use this Plug-In as an insert effect in any mono or stereo channel of your VST host. Don't overdue saturation effects on a whole mix. It's a good mixing strategy to apply a little here and there where saturation is actually helpful.

Quick reference

Dial in lower mid frequency boost by turning the knob clockwise. The meter surround- ing the knob indicates a rough overall volume estimate.
While being in the green area the added distortion should not be that sensible in most cases. However, on material like e.g. Acoustic piano recordings this will always be sensible and you might not want to use a device like this in such situations.

If the device is feed by an already hard driven signal (volume wise) it might not nec- essary to dial in more saturation by turning the knob.

NastyVSD

Overview

'NastyVSD' – a “virtual summing device”.
At a glance:
"NastyVSD" is a kind of "summing device" effects simulator and features some of the effects which can appear while going outboard (out of a Digital Audio Workstation (DAW)) and receiving a analog summed stereo mix back into the DAW. When doing this, "safety limiting" or even AD clipping is performed, so this is included in this simu- lation as well. When driven hard (aka abused), this could be used for maximising pur- poses as well.

Example Applications

- adding easily some “mojo” to recordings and mixes
- improving presence and room perception
- adding more sonic grip and warmth to thin sounding audio sources
- limit or hard-clip audio

Tech notes

- 4x oversampled input stage
- easy to use limiter
- minimum latency processing
- reasonable CPU usage
- Plug-in integration is done with Synthmaker software
- performance crucial parts are written in assembler or optimized by hand
- completely SSE optimized

General usage tips

Use this Plug-In as an insert effect in any stereo channel of your VST host.

Assure the 'IN' switch is in upper position (so LED is lightning red). If it's grey drag that switch into upper position. This toggles the overall operation (on/off).

Tip: *Use the 'OUT' knob to adjust the overall output level to equal levels and then use the 'OFF-ON' switch for convenient A/B testing (by switching from 'OFF' positon to 'ON' and back).*

NastyVSD is easy to handle, but there is just one thing to understand:

The signal flow is from left to right through the interface. Increasing the 'DRIVE' of the input stage (leftmost) also increases the internal signals volume at the point when leaving that stage and entering the limiter. This can be compensated with the 'GAIN' knob in the middle. In the same way the 'OUT' knob to the right handles the amount of hard-clipping, since the clipper is the very last stage in front of the Plug-Ins output.

Note: Opposed to version 1.0 of this software the internal routing slightly changed. In 1.0 there was the input stage followed by the limiter or clipper (switchable). Since version 1.1 there is a fixed routing as follows: input stage -> limiter -> clipper. Each tage is controlled by it's level dial.

So, if you want e.g. judge the saturation effects just when hitting the input stage you should lower the gain of the limiter which can introduce certain amounts of saturation as well.

Quick reference

(From left to right)

DRIVE: Increases or decreases the signal level while entering the input stage saturator
INPUT STAGE VU: Shows the signals volume performance at the ouput of the input stage
GAIN: Increases or decreases the signal level while entering the limiter or clipper
SLOW-FAST: The release time of the limiter. 'SLOW' is 300ms and 'FAST' is 10ms.
(LIMIT-CLIP: Switches between limiter and clipping mode) – removed with v1.1
LIMITER VU: Shows the signals volume performance at the ouput of the limiter/clipper
OUT: Increases or decreases the signal level while while leaving the device
ON-OFF: Switches the device on or off

NastyCS

Overview

'NastyCS' – a character channelstrip.

At a glance:

'NastyCS' features the very best "nasty" things coming from this Plug-In series and additionally two mid frequency EQ's originally developed for the BootEQ Plug-In as well as high- and low-pass filtering.

Example Applications

- adding easily some "character" to recordings and mixes
- frequency shaping with musical sounding EQ curves
- adding more sonic grip and warmth to thin sounding audio sources
- limit and saturate audio

Tech notes

- 4x oversampled output stage
- easy to use limiter
- minimum latency processing
- reasonable CPU usage
- Plug-in integration is done with Synthmaker software
- performance crucial parts are written in assembler or optimized by hand
- completely SSE optimized

General usage tips

Use this Plug-In as an insert effect in any stereo channel of your VST host.

Assure the 'OUT' switchs LED is lightning red. If it's grey click on it. This toggles the overall operation (on/off).

Tip: *Use the 'OUT' knob to adjust the overall output level to equal levels and then use the 'OUT' switch for convenient A/B testing (by switching from off to on and back).*

NastyCS features some “dual knobs” which are handled in this way:

This type of knob is actually two knobs, an inner and an outer one. The inner and outer one can independently dragged by clicking on the inner or outer part of the knob and dragging the mouse vertically up or down.
A symbol beneath the knob indicates this knob type and also shows which function belongs to the inner and the outer knob.
Example: The leftmost HP/LP knob is a dual knob where the HP filter is tied to the inner knob and the LP to the outer knob.

The signal flow in the output section:

(EQed signal) --> 'DRIVE' --> (internal saturator) --> 'LIM' --> 'OUT' --> level meter and output said

Quick reference

(From left to right)

HP/LP: High- and low-pass filter. To pass all frequencies through assure that the inner knob is leftmost and the outer one is rightmost (like shown in the picture above)

LF+/LF-: The Lowend EQ in boost/cut design: Inner knob boosts the lower frequencies while the outer cuts

GAIN/FREQ: The parametric mid filters – the inner knob decreases or increases the frequency (Zero position is middle/top like shown above) while the outer knob selects the frequency

HiQ: Switches into high Q mode of the EQ resulting into a steeper curve. Extremely useful to eliminate just certain audio artefact's with “notching”

HF SHELF: Boosts the high frequency audio content

DRIVE/OUT: The switch left to 'DRIVE' activates the saturator and now the inner (red) knob is active and determines the drive of the signal. The switch right to 'OUT' activates the device and the outer knob which sets the outgoing volume

LIM: Activates the limiter

FAST: Sets the limiters release time to 10ms (instead of 300ms)

BaxterEQ

Usage tips:

- *Use the 'IN' dwitch to toggle the plug-in on/off for A/B comparisons*
- *Use <ctrl> + mouse left click on a knob or switch to restore default position*
- *Use <shift> + mouse left click on a knob to fine adjust values*
- *Use this plug-in as an insert effect in any (stereo) channel of your VST host*

BaxterEQ at a glance

BaxterEQ – transparent mastering and mix buss shelving EQ

Finest tonal sweetening and finishing which always stays true to the source
- natural and accurate bass response
- authentic analog style HF curve rendering
- smoothest shelving operation

Perfectly suited for the mastering chain
- stepped controls throughout for repeatability and matched channel operation
- full dual channel layout
- full mid-side encoding support
- per channel level control for easy A/B match

Artifact free technical design
- low ripple and distortion filter implementations
- 64bit floating point internal processing
- oversampled for superior impulse response

Meticulously selected frequencies
- Baxandall shelving filters
- LF @ 74, 84, 98, 116, 131, 166, 230 and 361 Hz
- HF @ 1.6, 1.8, 2.1, 2.4, 3.4, 4.8, 7.1, 11 and 18 kHz
- 2-pole Butterworth filters
- LC @ 12, 18, 24, 30, 36, 43 and 54 Hz
- HC @ 7.5, 9, 11.1, 12.6, 16, 21, 28 and 40 kHz

In this device, an additional analog signal path emulation provides some subtle but precious stereo imaging improvements.

Reference

From left to right:

M/S switch – turns the internal mid-side encoding on or off
CUT – sets the frequency of the low-cut filter option
SHELF – sets the frequency for the LF shelving filter
Link switch – in upper position both channels LF controls are linked
LF – sets the amount of shelving operation for the LF shelf
HF – sets the amount of shelving operation for the HF shelf
SHELF – sets the frequency for the HF shelving filter
Link – in upper position both channels HF controls are linked
CUT – sets the frequency of the high-cut filter option
VOL – per channel output volume adjustment
IN switch – power on/off

Basic workflow

For easiest workflow it's recommended to always start in L/R mode, link both channels control sets and apply overall shelving and cutting as needed.
Then, unlink both control sets, switch to mid-side mode to refine EQing per channel and adjust the output volume control per channel as well.

The mid-side technique

The mid-side (M-S) stereo technique is one of the two formats of “intensity stereo,” that is, stereo in which spatial localization is determined by the differences in the intensity of a sound wave as it arrives in phase at a coincident pair of microphones. Intensity stereo relies completely on the dir- ectional characteristics (polar patterns) of the microphone pair to produce this effect, since only intensity differences and not phase differences exist between the channels for any single source arriving at a coincident pair.

(Source: "M-S Stereo: A Powerful Technique for Working in Stereo" by Wesley L. Dooley and Ronald D. Streicher)

The most common situations where M/S (aka M-S aka mid-side) techniques are getting applied are:

- during the recording process when mid-side microphonie is utilized
- on the 2bus during the late mixing or mastering stage
- in audio restauration situations

I'm skipping the mid-side microphonie stuff here and just recommend "A More Realistic View of Mid/Side Stereophony" by Trevor Owen de Clercq which is available online at http://www.midside.com/pdf/nyu/masters_thesis.pdf.

While the M/S signal is obtained in a natural fashion during microphone stereophony, in most other cases stereo (L/R) encoded signals are available. In this case and to obtain the advantages of M/S a conversion is necessary.
The conversion is dead easy and is accomplished via a sum-and-difference matrix network, where typically the mid signal is the sum (M = R + L) and the side signal is the difference (S = R – L). The other way around is that simple as well and there are plenty of tools available which handle this stuff for us in the digital audio workstation.
Once a stereo source is encoded in M/S, the door is open to treat mid- and side content individually. For example, this allows for selective correction of some problems encountered on location such as out-of-phase low-frequency noise from the environment. In this case the side channel, which contains the majority of this information, can be passed through a high-pass filter to reduce such unwanted low-frequency content, and this can be done without any alteration of the mid content.

Some other typical mixing/mastering targets ideally achieved in the M/S domain are:

- assuring mono compatibility
- stereo widening and increasing depth perception
- attenuating or emphasizing the signals room information
- increasing intelligibility of voice in a mix

This is accomplished mostly by performing alterations of both channels frequency and dynamic response. The tools to be used just need to have separate controls per channel as long as M/S encoding/decoding is done externally before and after the plugin, other tools already offer internal M/S handling to make things easier: The BaxterEQ is an example for a M/S frequency shaper and Density MKII is an example for a M/S dynamics compressor.

NastyVCS

Usage tips:

- *Use the power switch on the right side for handy A/B comparisons*
- *In the toolbar in the bottom of the interface the IN switches must be active to run each section - use them for A/B comparisons as well*
- *Use <ctrl> + mouse left click on a knob or switch to restore default position*
- *Use <shift> + mouse left click on a knob to fine adjust values*
- *Use this plug-in as an insert effect in any mono or stereo channel of your VST host*

Overview

NastyVCS – Virtual Console Strip.

Inspired by the smooth dynamic and tone shaping capabilities of some high-end mixing consoles and channel strips, this plug-in implements the most distinctive and much appreciated sonic effects generated by these devices:

- filtering and equalizing
- preamp style saturation and phase adjustments
- opto-electric compression

Functions at a glance

- performs gentle audio dynamic treatments
- masters difficult to handle audio material in a musical fashion
- shapes frequency and phase response
- adds extra harmonics and saturation effects
- controls outgoing audio peaks

Plug-in specification

- Win32 / VST compatible
- state-of-the-art digital signal processing
- performance-critical parts are written in assembler
- completely SSE optimized

Getting the most out of it

Please read the following chapters to get the most out of this device. Learn how to efficiently set and combine each section:

- obtain some effective tips on getting the most out of the presets
- understand the basic workflow of this device
- learn how gain-staging is handled
- take advantage of the sidechain and routing options
- see how dynamic treatments can be split across compressor, limiter and satu- rator
- understand how EQ and filtering is implemented

Quick reference

From left to right according the graphical interface :

Nª	GUI	AUTOMATION	DESCRIPTION
1	IN	Preamp In	Turns the preamp stage on
2	SAT	Preamp Sat Level	Sets the amount of preamp saturation
3	IN	Preamp Input Level	Sets the audio input level (post saturation)
4	IN	Filters In	Turns the two filters on
5	SC	Filters to SC	Routes both filters into the sidechain path of the compressor
6	HP	Filters HP Freq	Sets the HP filter frequency
7	HiQ	Filters HP HiQ	Switches into high Q mode (24dB per octave instead of12dB per octave)
8	LP	Filters LP Freq	Sets the LP filter frequency
9	HiQ	Filters LP HiQ	Switches into high Q mode (18dB per octave instead of 12dB per octave)
10	EXT SC	External SC	Takes the compressors sidechain signal from audio input 3 and 4 (instead of the main channel 1 and 2 in's)
11	IN	EQ In	Activates the EQ section
12	POST	EQ Post Comp	Routes the EQ section behind the compressor
13	BOOST	LF EQ Boost	Low frequency boost
14	CUT	LF EQ Cut	Low frequency cut
15	FREQ	LF Freq	Low frequency center
16	HiQ	LF HiQ	Switches into high Q mode (results in steeper EQ curves)
17	GAIN	LMF Gain	Low mid frequency gain amount
18	FREQ	LMF Freq	Low mid frequency center
19	HiQ	LMF HiQ	Switches into high Q mode (results in steeper EQ curves)
2ß	GAIN	HMF Gain	High mid frequency gain amount
21	FREQ	HMF Freq	High mid frequency center
22	HiQ	HMF HiQ	Switches into high Q mode (results in steeper EQ curves)
23	AiR	AiR Gain	Turns in the high shelf
24	8k	-	Sets high shelf to around 8k
25	12k	-	Sets high shelf to around 12k
26	17k	-	Sets high shelf to around 17k
27	COMP	-	Sets metering to display compression gain reduction
28	LIM	-	Sets metering to display limiter gain reduction
29	OUT	-	Sets metering to display output volume
30	P-IN	Phase In	Turns the phase tool on
31	INV	Phase Invert	Switches the polarity of the audio signal
32	90/180	Phase 90/180	Switches between 90 and 180 degree phase shift
33	PHASE	Phase Shift	Adjust the phase shift from higher to lower frequencies
34	IN	Comp In	Turns the compressor on
35	+12Db	Comp Input	+12dB Boosts the compressor incoming signal 12dB
36	COMP	Comp Level	Sets the compression level (drive)
37	GRIND-PRESS-SLACK	Comp Attack	Sets the overall attack time characteristics
38	SQEEZ-THRUST-RELAX-SOFT-L.A.	Comp Release	Sets the overall release time characteristics
39	MAKEUP	Comp Makeup	Gain adjustment after compression
41	IN	Limiter In	Switches the limiter on
42	FAST	Limiter Fast Release	Selects a faster release time characteristic
43	LIM	Limiter Level	Turns the limiter transfer curve from 1:1 to ∞:1
44	OUT	Output level	Output level
45	-	Power	Turns the whole plug-in on/off

Basic operation and advice

Use this plug-in as an insert effect in any stereo or mono channel of your VST host. It can be operated both as a mono or stereo plug-in. If your host supports sidechain routing the external sidechain feature can be used as well.

Make sure that the power switch on the right side is in on position now (lightning). Make also sure that in the toolbar in the bottom of the interface each sections IN switch is active (lightning) if this section is needed.

For example turn on the IN switch for the compressor (straight below the COMP knob) and dial in some compression with COMP. Compensate for volume loss with the MAKEUP dial. The attack and release time switches can be dragged up and down to select a specific program.

Use the IN switches for A/B testing.

While the IN switch enables or disables a whole section, setting the SAT, COMP or LIM dial to 0 bypasses the according effect but not the whole section.

Some tips on using the presets

Rather then offering a variety of presets for specific mixing situations NastyVCS features some templates which you can use as a starting point and adjust to your needs. There are some “Preamp simulation” programs, some for “virtual summing” and some other dedicated to certain program material or mixing tasks.

Always adjust the audio input level (which “drives” certain components of the plug-in) and the output level (for A/B tests).

Understanding gain-staging in NastyVCS

The audio signal flow in NastyVCS is basically from left to right through all the components as shown in the interface (with few exceptions as discussed in Chapter 3.2).

Beside that, there is just one single concept to be understood to know how gain-staging is actually working in NastyVCS internally. This affects the SAT, COMP and the LIM section and works identically with each of them and so just one of them is explained here.

Gain-staging in each of these sections is basically working from the bottom to the top which means:

1. The IN switch determines if the whole section is in or not and therefore if any level changes are applied or not.
2. Next comes the lower of the two knobs which can alter the audio volume as a side effect (SAT in this case).
3. Note: If this knob is set to 0 this is a true bypass of the effect (but not of the whole section).
4. The last step is the knob above (IN in this case) which alters the audio level after the main effect (SAT in this case) has been applied.

The concept of a true bypass as hinted in step 3 allows to use one section for just gain-staging purposes and not applying any other effect. For example if you don't want to apply any compression effect but want to drive the signal upfront the limiter just use the compressor stage and set the COMP dial to 0. Compression is now entirely bypassed and you can simply use the MAKEUP dial to adjust the audio level.

Advanced Usage

Side-chaining and routing

Basically the audio signal in NastyVCS flows from left to right through all the compo- nents as shown in the interface. There are two exceptions for this, one is the sidechain signal flow and the other are the routing options.

The sidechain signal is basically the audio signal which is presented to the compressor to calculate it's gain reduction amount from. In NastyVCS this can be obtained from the main audio inputs (channel 1 and 2) but can be selected from channel 3 and 4 as an option by switching EXT SC (external sidechain) on. Your host must support this, of course.
In both cases the filtering stage (HP and LP) can be switched to appear in the sidechain path and not the main audio path anymore. This is done with the SC switch in the filter section.

By default, the EQ section is in front of the compressor. This can be altered by turning on POST in the EQ section. The EQ is then located between compressor and limiter.

Saturation and phase alignments

NastyVCS offers a dedicated input stage which allows simple input gain control but features crunchy “pre-amp” saturation as well. The amount can be dialed in to taste and allows very first peak treatments ranging from subtle to nasty (sic!).

Preamp stages usually are affecting both, saturation and phase alterations but in NastyVCS this is separated to allow way more and precise alignments of the audio signal's phase. The phase section is located left of the compressor and the combination with the compressor and/or the saturator then is the ticket towards “preamp style” or “opto comp” audio signal path coloring.

The input saturation features harmonic distortion and is a good alternative or complement to limiting. Example: on difficult to handle bass tracks the input stage can do wonders the limiter can’t. Finally, the combo of both can be a real killer in some mixing situations.

The phase tool basically combines all-pass filtering in a special setup plus some additional controls. The basic phase shifting shows up as in the diagram below (90/180 phase adjustment engaged). It features an additional polarity switch (also referred to as phase inversion, hence the INV switch label) and a phase frequency center knob. The diagram shows different PHASE knob positions ranging from 0 (right-most curve) to 12 (left-most curve).

The interesting thing about this approach is that even if some rather constant phase shifting approaches (which are possible in the digital domain) might appear to be more reasonable or accurate in theory the method still maintains a more interesting or musical compensation at the end (since phase alteration in the analog domain does not appear as theoretical ideal).

Such alignments can be used for rather “artistic” mixing treatments and audio signal coloring as well. Just as a simple example, adjusting it properly to the upper frequency range one can easily achieve some serious amounts of transient smearing or, applied to the lower frequency part, the bass range can be decoupled to some extend (some bass enhancers are taking advantage of such effects as well).

Compression and limiting

The heart of NastyVCS is the entirely new build opto-electric style compression unit. This type of circuit designs do have a significant and highly program dependent behaviour and this is mainly due to adaptive release time characteristics, the typical compression transfer curves and the overall frequency dependency and non-linearities (have a look to chapter 4 for more details). Result: smoothness even on difficult to handle audio material.

An external sidechain option is available for the compressor (engaged by EXT SC) and with SC turned on the filtering section can be routed into the sidechain. If you prefer the EQ being behind the compression stage then engage the POST switch in the EQ section. Last but not least, the sidechain signal can be boosted with the +12dB switch to easily increase the compression amount.

While creating this compressor, major efforts has gone into designing the specific attack and release time characteristics which are available as fixed programs: three programs for setting the attack and five for release time behavior. Those programs cover a really useful range from faster to slower timings but since everything acts program dependent no precise timing information is specified. They are sorted timing wise from fast (top) to slow (bottom).

This compressor maintains punch in almost every situation and you can’t really go wrong with it – it’s a stellar but easy to use mixing comp. If you are in the need for more explicit dynamics treatments with NastyVCS then simply choose the limiter. Placed right to the end of the signal path, the limiter gives the final transient control and can smoothly be dialed in from zero to 100%. Set to 100% it works as a true and accurate brickwall safety limiter to prevent peaks to go further above 0dBfs.

As with each and every other NastyVCS component, the limiter does not need any lookahead information and so even limiting is available for latency free tracking. Two timing options (selected by the FAST switch) can be chosen, complementing the compressor for final management of audio transient information.

The internally calculated amount of gain reduction can be displayed visually with the COMP and LIM option right below the vertical VU style meter. On top of that, the lim- iter features a little clipping indicator (small LED right beside the LIM label) which roughly indicates clipping events.

EQ and filter

Similar to the EQ's and filters in BootEQ mkII, NastyVCS stays on the musical (and not the surgical) side of the audio source. It is more a coloring toolbox (especially in the combination with the PHASE option) rather than allowing to shape audio beyond recognition. Some classic technical principles have been carefully selected and replicated and the overall combination and attention to detail makes NastyVCS stand out in the crowd of equalizers today. Given the specific curve selections you might notice its slight “old-school” attitude when mixing through NastyVCS.

To get an idea of this and the overall philosophy of NastyVCS let's have a closer look at some of the technical designs starting with the boost/cut style equalizer. Boost/cut style equalizers are typically obtained from EQ's working in parallel configurations and “taking advantage” of certain effects coming from their interacting phase response. In NastyVCS I did not only want to have a basic boost/cut EQ, I wanted to have the boost curve from one of my favorite bass EQ's, combined with the cut option to allow those classic boost/cut curves. This ended up in a way more complicated technical design which actually combines some serial and parallel filter configurations.

Likewise with the mid frequency bell type EQ's their boost and attenuation behavior is calculated individually, using different algorithms.
Following the same concept, the HP in the filtering section is implemented as a straightforward 12 or 24dB butterworth filter while the LP features a different and smoother design which avoids warping near the Nyquist frequency and offers gentle high-end treatments.

Last but not least the special AiR shelving EQ resembles the top end curve of a specific mastering EQ but also adds that certain slight “dip” below the boost which is so well-known from another type of (musical sounding) mixing EQ.

Putting it all together

NastyVCS really shines when all its tools are properly combined. Condense the dynamics in some small amounts right in the input stage, apply some smooth compression without affecting the punch/transients and then let the limiter eat some peaks at the output. Combine saturation and phase shifting to obtain some fancy audio signal coloring or just use the phase option for pure alignment tasks during recording. Gently limit the audio frequency range with the smooth filters or use them in the sidechain while adding some musical texture with the 'old-school' equalizer curves.

The beauty of opto-electrical compression

Opposed to VCA, Variable-Mu or FET based approaches, opto-electrical compression takes advantage of using a light-sensitive resistor and a small light emitter (a LED or electroluminescent panel) to obtain a gain reduction voltage in the sidechain path. This technique is well-known to add some smoother gain riding characteristics to the signal because of the specific attack and release response which comes from the inertia and inherent memory effect of the photoresistor element.

Program dependency

Typical classic opto-electrical circuit designs do have a significant and highly program dependent impact on the processed sound and this comes mainly from three factors:

1. The adaptive release time characteristic which gets faster at higher compression activity and can be fairly long when leaving the compressors duty cycle.

2. The specific compression transfer curve which features soft-knee characteristics by nature and limits the dynamic range.
3. The frequency dependency and non-linearities impacting the actual behaviour of things like the compression transfer curve.

In addition, opto elements do have an inherent lag time in their attack response which is typically not fast enough to catch short transients but adds up to an overall smooth gain riding impression.

Light and shadow

Opto compression really shines on overall adaptive and smooth gain riding purposes such as for vocals and solo instrument performances, whereas faced with rather complex program material it can easily sound a bit quirky. Arguably that's why it is less ideal on the mix-bus, at least if you need bigger amounts of gain reduction.

In the digital domain the effects of the opto element are artificially modeled anyway, so these drawbacks can be avoided and both frequency dependency and non-linearities can be applied nicely to full program material.

About audio signal coloration

In this comprehensive article some deeper explorations and explanations on this topic are given and at the end a brief but handy definition about audio signal coloration is proposed. Some tips on mixing can be obtained here as well and – by the way – some myths about equalizing audio in the digital domain get busted.

But first let’s have a closer look into a different domain, the domain of digital image processing. In digital image processing, the fundamental color impression of an image is actually changed by performing some proper DSP maths on parts of the color spectrum of the image as shown in the example diagram above. Typically, a digital image is encoded into a 3- or 4-dimensional color space (like RGB or CMYK) and then each dimension can be manipulated individually over the spectrum. This changes then the overall coloration (and other things like brightness or contrast as well). Quite similar, in the audio domain there are two dimensions over the frequency spec- trum which can be utilized to alter the audio color impression: the magnitude and the phase response and this is typically done by an equalizer or filter. We are not going to talk here about those drastic phasing effects which are typically introduced by time shift based effects such as chorus and delay. While the impact of alterations in the frequency magnitude response curve is quiet obvious, altering the phase response might be not (and is often mixed up with other EQ side effects like resonance or ringing of a filter).
So, how does a certain phase alteration actually affect the sound perception of the audio? Of course this depends on the real frequency where the phase alteration (aka phasing, phase shift, phase drift, phase warping, phase distortion) occurs but for the sake of simplicity lets first have a look at the rather general effect caused by an overall and continuous phase shift: Lets assume we have an effect which introduces a continuous phase shift over the entire frequency range but changes nothing else (which can be performed by an Allpass filter – see the figure above). Now we have three scenarios, depending on the amount (degree) of the phase shift:

1. Slightest phase shift: Human ear does not perceive anything and thus can’t judge any better or worse sonic quality.
2. Some amounts of phasing: The phasing, which causes some drift of higher frequencies in time, can now be perceived by our hearing. This slight displace- ment in time could be perceived as to be a more sound, less edgy and harsh audio quality and even to have more room / depth impression. (Hint: this are qualities some people hear and associate to the rather positive properties of analog audio processing).
3. Larger phase drifts: Larger displacements in time increase this effect and can completely destroy the transients. The signal is perceived as being washy and roomy now and lacks definition which is not desired in most cases (but can be useful e.g. as part of reverberation processors).

In digital reality, phase shift is not introduced that much over the entire spectrum by our commonly used DSP mixing effects such as phasing EQ’s but has a rather local effect and is not that easily detected by ear since the phasing effect is being concealed by the frequency magnitude change of the EQ. This holds true at least as long as gentle and broad magnitude changes are performed. That said, if rather deep and steep changes are made then also larger and maybe unwanted spectrum displacements are introduced by such an EQ.

This may lead to some serious issues when for example on each and every track most signal resonances were removed by such steep filtering effects which is a com- mon misconception in mixing audio. It leads not only to rather flat and boring signals but also introduces significant phasing issues as described above as long as no lin- ear phase EQ is used (which introduces other problems and is not discussed here) and as an overall result the mix gets fluffy and lacks definition. As a side note, this also shows that the prejudice that cutting is always preferable to boosting other frequencies is an urban myth.
In some cases it might be considerably better to gently boost the desired frequencies instead of deeply cutting some unwanted ones and the simple “garbage in, garbage out” law applies here too: If that much and rather deep cutting or filtering in general would be necessary on such a signal then it’s probably better to try to fix this by changing the source, the recording situation or the arrangement. A good excercise is to set up some sound sources plus arrangement where (almost) no EQing is necessary during the mix. As an added sugar, such well-selected/recorded and arranged sound sources typically lead to a way better and much more natural loudness performance in the end. But back to topic.

So, is this phasing really bad and has to be avoided in any case? As hinted earlier, applied in slight doses, phasing can introduce a very nice and pleasant audio signal coloration and is part of the sound that we typically associate with high quality audio processing in the analog domain. In DSP land, these kinds of effects can be used to the audio engineer's advantage as well by simply applying phasing in the right amounts and in the right place of the spectrum. This is implemented e.g. in some audio enhancer circuits which are introducing dedicated phase shifts aimed at the spectrum or specific to the loudness performance of the audio signal.

Even some digital compressors are utilizing this. This answers yet another interesting question - can we also introduce audio signal coloration just by using plain dynamic effects? The short answer is: Yes we can! This is rather obvious and can easily proved by an audio analyzer in many,though not in all cases. Some compressors e.g. introduce gain reduction dependent phasing which can subtly change the color impression and of course a true multi-band compressor is able to perform drastic frequency magnitude changes for obvious reasons.

But even if the dynamic processor does not alter the frequency or phase response in a direct fashion it can alter the perceived spectrum just by having implemented a frequency dependent sidechain treating parts of the spectrum differently from others in respect to their loudness performance. This is also true for transient processors in general when transient information is typically associated to (and treated in) specific frequency ranges.

The phrase “audio signal coloration” could simply be seen and understood as “affecting the perceived tonal spectrum” no matter which phenomena or method actually caused it. A phasing EQ, when properly applied, is a good way to pleasantly color the audio in both dimensions, frequency magnitude as well as phase response. A linear phase EQ colors the audio too but just in one single dimension. Other processors such as compressors or enhancers can potentially take advantage of audio signal coloring as well, not even mentioning the time shift based effects such as chorus or delay.

Judging saturation effects

There are quite some misconceptions around on how to judge a saturator's sonic quality, here are some tips to avoid the most common pitfalls:

1. A good saturator does not appear as distortion in the very first place. Firstly it just saturates incoming audio signals which means that at a similar RMS output level it simply reduces the peak performance (which results in a smaller “crest factor”).

2. This immediately implies that you need a RMS meter in your output chain to compare different saturation settings or devices to another. Basically this is the same for comparing limiters or maximizers.

3. Distortion is a side-effect which typically occurs at higher saturation levels. It can have different sonic qualities, e. g. due to the frequency distribution of distortion which makes a huge difference to human hearing and whether the effect is perceived as gentle or not.

4. Don’t rely on a simple spectrum analyzer here, it does not know anything about the concept of being “gentle” or not.

Summary: Always assure equal RMS output levels and then use your ears.

preFIX

preFIX – getting those alignments done.

preFIX is a pre-mixing and audio alignment tool which typically takes place upfront the mixing process. It provides a clever tool set to clean-up, fix and align audio tracks (typically taken from recordings) concerning overall frequency correction, phase alignment, spatial stereo field corrections and routing. It contains a complete gate/expander solution with a dedicated and comprehensive sidechain filtering path as well.

Highlights

- smoothest audio frequency filtering
- comprehensive gate and expander audio treatments
- detailed phase corrections
- easy routing and stereo imaging changes

Plug-in specification

- Win32 / VST compatible
- state-of-the-art digital signal processing
- performance-critical parts are written in assembler
- completely SSE optimized

Features

- Main audio path filter
- Baxandall style shelving filter with pristine audio quality
- Smooth Butterworth high- and lowpass filter with switchable characteristics (12 and 24dB per octave)
- Both are oversampled and match their analog model curve behavior
- Sidechain path EQ and filter
- Whole spectrum “tilt style” balancing filter with adjustable center frequency
- Smooth Butterworth high-and lowpass filter with switchable characteristics (12 and 24dB per octave)
- A dedicated parametric peaking EQ
- Internal/external sidechain switchable
- Sidechain listening option
- Gate/Expander
- Adjustable threshold between -80 and +6dBFS
- Freely adjustable knee from hard- to soft-knee behavior up to 1:2 downward expander mode
- Range limiting option (floor)
- Channel link switchable
- Envelope follower section with attack, hold and release control
- Attack timing features console style peak timings as well as two rms modes
- Additional gate pre-open timing option
- Phase alignment
- Analog style step-less signal phase corrections
- Detailed options for adopting phase response curve regarding polarity, frequency center and width
- Additional digital signal delay option
- Phase control switchable to channel 1, 2 or both
- Advanced control
- Detailed output channel routing with six different modes
- Stereo image rotation option visually supported by a goniometer
- Output level control with special mono mode to mix in channel 1 and 2

Overall signal path flow

The overall main audio path routing is not strictly from left to right through the interface but goes according the following order:

1. EQ
2. Gate
3. Phase alignments
4. Output routing and stereo imaging
5. Output level control

The routing for the SC signal path takes its input directly from the plug-ins inputs 1+2 (internal sidechain) or 3+4 (external sidechain) which are routed then through the SC EQ section and directly into the gate/expander.

If the sidechain listen function is activated then the gates audio output signal is dropped and replaced by the SC EQ output. All processing afterwards in the main audio path (phase alignment, routing etc) still applies!

Note: Each section contains a separate on/off switch on the bottom which must be powered on to run each module.

Advanced

The modules explained

Main audio path filters

There is one EQ/filter section fixed in the main audio path
(the left most one).

That one is for the main audio path only and is a modern Baxandall EQ adoption but with freely adjustable frequency and gain settings. It also offers smoothest Butterworth high- and lowpass filters which can be switched between 12 and 24dB per octave characteristics.

This whole section is oversampled for best quality and pre- cise curve match even in the highest register. Its an incred- ibly good sounding unit and you can't harm any incoming audio by dialing in these (shelving) filters. All frequency readouts are in Hz.

SC path filters

Another EQ/filter is permanently located in the SC path (titled SC).

This one only works on the gate/expander sidechain path and offers a 12/24dB HP, a peak/notch filter, the “tilt style” whole spectrum balancing filter (with additional possibility to adjust the center frequency) and a 12/24dB LP.

This EQ lane can be listened to with the little speaker sym- bol switch. External sidechain is also fully supported by the EXT/INT switch. Note that for external sidechaining your plug-in host must route a proper audio sidechain signal to input 3 and 4 of this plug-in.

The gate/expander

Adjusting the transfer curve with the 3 sliders:
- TRH sets the threshold level at which gating/expanding should begin.
- RANGE limits the range of the gating/expanding (also known as "floor"). In bottom position this function is off (no range limiting).
- KNEE sets the curve characteristic from instant/hard (0%) down to a smoother knee curve. In bottom position the curve represents a 1:2 downward expander curve with smooth soft knee.

Adjusting the timing:
- ATT: switches between five fast peak signal detection timings (0.1 – 1.5ms) into two different RMS detection methods (5 and 15ms). The peak timing behavior relates to some console channel gates. The RMS timings can be useful to easily avoid fluttering of the gate on a micro dynamics level (e.g. when gating toms in the release tail) or to better catch a snare in a whole drum mix for example.
- HOLD: Sets the hold time for the envelope.
- REL: Sets the release time for the envelope.
- PRE: Adds a gate pre-open control so that the gate opens earlier then the audio event actually occurs.

Additional options:
- LINK: Links both channels in the sidechain path.
- 40/80: Sets the threshold level range from 0...-40dB down to -40...-80dB

The scope section

In the scope section there is a Goniometer included which represents the stereo field imaging. The display can be adjusted with the SIZE and GLUE knobs which do not affect the audio itself!

There is a selector knob OUT which allows to select from six different output routing modes. Afterwards in the audio path, the ROT knob rotates the audio signal in the stereo field.

The six routing modes explained:

Stereo Ch1 input routes to Ch1 output and Ch2 input routes to Ch2 output
Mono The mono content of Ch1 and Ch2 are routed to both, Ch1 and Ch2
Mono* Same as mono but the mono conversion is applied after output level faders
Swap Same as stereo but with swapped channels
1->1&2 Routes Ch1 to both, Ch1 and Ch2
2->1&2 Routes Ch2 to both, Ch1 and Ch2

Output level control

To the right side of the panel two faders are provided for output volume control per channel which is displayed then in the final volume meter. Both sliders can be linked.

Note: In Mono* mode both sliders are active on each channel upfront converting the output to mono.

Phase alignments

The phase alignment tool basically matches an analog phase shifter with some additional digital options.

This section provides:

- A channel switch: Sets the phase operation to Ch1 or Ch2 only or both.
- DLY: Introduces a digital signal delay up to 4ms.
- PHASE (the knob on the right side): Alters the phase transition around its center.
- 0/INV: The polarity switch.
- 90/180: Sets the width of the phase alteration.
- LO/HI: Sets the center of the phase transition.

Tips and tricks

Improving the stereo field perception

Beside the stereo field rotation option in the scope section the phase alignment is the ticket. Just apply this alignments to just channel 1 (or 2) with the switch on the top. Dial in some subtle phase alterations and check stereo field behavior visually with the scope. Check for mono compatibility with the mono routing option.
Works great with vocal groups or rhythm guitar sections!

Focusing the spectrum and tonality of all tracks

Applying high- and low-cuts to each and every track is a well known rule of thumb before starting mixing. Though, this should be considered with care especially for acoustic recordings. Some audio lowend content might actually be the frequency body of an instrument and the HF typically contains the overtone structure. So, choose all cuts with care. preFIX provides two filters dedicated for this task in the first EQ section.

Dealing with gate release tail distortions

Might be a difficult task if the audio content is sensible to distortion artifacts, e.g. as with release tails of toms. Utilize the HOLD timing option of the gate in this case or tweak the sidechain EQ to better focus the detector for the gate. Soft knee is also a good option.

Dealing with fast gate attack timings and distortions

If fast attack timings are needed but attack transition becomes a problem try to utilize the pre-open feature of the gate to move the gates opening upfront the audio event itself.

If the track needs some “balls”

Then the EQ is the ticket: Use highpass and shelving filter altogether to boost the LF department while cutting the very lows the same time.

If the track needs some “air” or “shimmer”

Just dial in the pristine sounding Baxandall HF shelving filter – you can't go wrong.

Align stacked recordings

The phase section is the right place to look: Just apply this alignments to just channel 1 (or 2) with the switch on the top of the section.

Getting those tight (electronic) drum tracks

Use the gate, Luke! Use it on the kick, the snare, the hi-hats, on everything. Use it on single tracks and the drum sub-groups. Set each tracks focus with the main EQ filter section on top. Always check for mono compatibility.

Mixing a stereo track into mono

Check out the mono* option for this task which gives you the opportunity to adjust each channels mixing level before everything is converted to a mono signal.

Workflow: separate track alignment from mixing

Start first to check all tracks and if mixing is feasible with them. Some might not match the quality and needs to replaced. Some just need some basic alignments. Do all creative decisions and treatments later during the mixing process. Mixing is always context dependent.

Alignment does not mean restoration!

If you prepare a track and something like “where is my damn 64dB/octave brickwall filter?” comes to your mind then this might be because your are not upfront any mixing process but are working on audio restoration instead. Or the sources just might be crap. Always remember: garbage in, garbage out.

Closing comments

preFix was developed together with quite a bunch of recording engineers. The specific technical design and feature combination found in this unique audio plug-in might not be self-explanatory in the very first place but I can just encourage to explore all the possibilities to discover how things are working nicely hand in hand. Some hints and applications are already given in the previous chapter.

Addendum

Compressor, gate and expander

Some might get confused sometimes when compressor, expander and gate are discussed and especially when concepts like “upward”, “downward”, “parallel” or suchlike are thrown in. Fortunately, things can easily be explained just by looking at the according transfer curves and as an added sugar some more sophisticated insights can be obtained es well.

The “ordinary” compression transfer curve which is used in most audio dynamic compressors is the downward compression curve somewhat similar to the image above where a 1:1 curve is shown for reference as well (and so in all other diagrams here). It basically works in a way that a signal level above a certain threshold – in this case roughly around minus 10dB - leads to an attenuation which is linear as well (ok, almost – and just in case of a hard-knee compressor) but with a lower amplification ratio. The so-called upward compression works the other way around: A signal above the threshold level remains unaltered level wise but those below are getting amplified to a certain degree now. This is shown in the diagram below where a threshold is set somewhere around +20dB:

Since the upward compression transfer curve looks quite similar to a *parallel* downward compressed one (which is an uncompressed dry signal mixed back into downward compressed one) some might guess that this is completely the same but actually it’s not. The difference lies in the fact that envelope curve based dynamic treatments are only partially applied to the parallel compressed signal and the other part remains completely unmanaged. This is different in the upward compressor: the whole signal will be shaped by the envelope curve and therefore the transient response is actually different.

Opposed to a compressor an expander features basically just different transfer curves as shown in the diagram above where a typical 1:2 downward expansion curve is shown: In this case, below a threshold at around -10dB downward expan- sion occurs which even lowers the already quieter signals. Upward expansion (not shown there) is just the other way around and preserves the signals below a threshold and amplifies the already louder ones. The downward expansion is often referred to when it comes to the so-called Gate/Expander device. This can easily be understood just by looking at the according transfer curves again which in this case shows that the classic gate curve is just a special case of the downward expander where the ratio becomes infinite:

Additionally, other concepts like soft-knee curves (already shown above) or processing range limiting (like a range control in a compressor or a floor setting in a gate) can be implemented just by proper transfer curve designs as well. This leads to the question, if one could implement just one single device for all that three purposes – compressing, expanding and gating – by simply utilizing the very same basic engine and just swapping the respective transfer curves. Well, in theory this idea might be attractive and sound but practise has shown that this does not lead to optimal results. All the other relevant aspects such as time or frequency dependent behaviour or even non-linearity is not that interchangeable between such devices and their specific application domains in general.

Density mkII

Usage tips:

● *Use the MAKEUP knob to level the outgoing audio and for handy A/B comparisons*
● *Level your audio input to the Plug-In to around 0dbFS to perform easy and best inside the plug*
● *Use <ctrl> + mouse left click on a knob or switch to restore default position*
● *Use <shift> + mouse left click on a knob to fine adjust values*
● *Use this Plug-In as an insert effect in any mono or stereo channel of your VST host*

Overview

Density mkII – smooth and versatile dynamic processing on the stereo bus.

This device ain't modeled after any specific outboard gear but rather incorporates some proven dynamic shaping approaches from the past, combined in a seamless fashion with some much more modern concepts in audio processing - the best of both worlds. Density mkII was primarily designed to work in a typical stereo audio group mixing situation or while summing and to glue all things together in a rather unobtrusive way. Yet it's capable of signal colouring but in a subtle and pleasant way and also can perform quiet different tasks very versatile.

At a glance

● perform ultra smooth 2bus compression
● master difficult to handle audio dynamics
● manage mid/side dynamic processing in a true two channel layout
● apply dynamic range adjustments easily

Plug-in specification

● Win32 / VST compatible
● state-of-the-art digital signal processing
● zero latency processing, no phase alterations
● performance crucial parts are written in assembler
● completely SSE2 optimized

Getting the most out of it

Please read the following chapters to get the most out of this device. See and learn especially on how to:

● see some effective tips on getting the most out of (boring) presets
● focus the actual gain reduction by simply limit the actual dynamic range of the compressor
● explore the exciting limiter mode which gives incredible dynamic control even on difficult audio sources and can master some typical leveling amp tasks
● see and hear how the RELAX feature retains punch in your mix
● use mid/side processing to your advantage and obtain smooth and balanced stereo mixes
● learn some cool tricks to make mid/side compression really easy by using the side-chain 'unlink' feature and other shortcuts

Quick reference

Large knobs:

RANGE – adjusts the actual gain reduction range from 100% (right) to around 5% (left most)

DRIVE – increases the gain of the audio signal in the side-chain path
TIMING – selects six different attack/release time combinations
MAKEUP – increases the audio level after compression

Small screws:

LINK – links the two channels in the side-chain path
DRY/WET – mixes compressed and uncompressed signal
STRICT/RELAX – the more to the right the more the timing parameters are relaxed

Switches:

ON/OFF – basic on/off operation
FILTER/OFF – turns the filter in the side-chain path on or off
INT/EXT – feeds the side-chain input signal from internal or external channels
COMP/LIM – switches between the two basic compression modes
LR/STEREO/MS – selects the channel operation and internal encoding

Meters:

VU – displays signal levels after
DRIVE GR – displays amount of gain reduction

Basic operation and advice

Use this Plug-In as an insert effect in any stereo or mono channel of your VST host. It can be operated both as a mono or as a stereo Plug-In. The best performance is obtained if the Plug-In is applied to a true stereo channel audio.

Assure the POWER switch is in ON position (power indicator is light- ning). If the indicator is grey click on the switch until it's lightning. This toggles the overall Plug-In operation (on/off).

Level your incoming audio so the needle of the VU style metering in the left section of the Plug-In clearly shows some movement. Dial in amounts of the 'DRIVE' knob to apply further compression effects to your audio.

Apply different compression timing constants just by using the TIMING dial. P1 offers fastest attack and release speed and P6 is slowest.

Check both compression modes: COMP and LIM. They both offer different and exciting dynamic shaping possibilities. Don't see LIM just as an ordinary signal limiter and you'll get rewarded.

Use the MAKEUP knob to adjust the overall output volume as needed and for handy A/B testing at equal volume levels.

Some quick tips on using and adjusting the presets

Explore the presets but always adjust them to your current mixing situation:

- Always adjust the DRIVE parameter to your current track or mix so that the actual applied gain reduction / gain riding actually matches the specific needs.
- Always adjust the MAKEUP parameter to your current mix for equal loudness situations during A/B listening.
- Special mid/side processing presets are labeled with M/S in it's name.
- 'NY' intends special parallel compression in 'New York” style.
- 'LA' refers to leveling amp task presets with longish exponential release curves and variable soft-knee curves – don't miss them out, they are excellent for e. g. vocal group tracks or mixes with overall large volume level changes over time.
- If a certain preset/setting alters the stereo image perception too much: Check and adjust the LINK screw, which in most cases does the job. If not, engage M/S mode for further tweaking the stereo image perception.

Reference

Compression modes

Beside the COMP compression mode, Density mkII offers a way cool limiting mode: LIM. Don't relate this to some brickwall limiter designs or such a like but just see this as another creative and powerful dynamic shaping option with different topology and sound.
While the COMP mode offers a rather hard-knee compression transfer curve with a gentle ratio between around 2:1 and 4:1 (and in feedback
topology), the LIM mode works in feed-forward topology and features a super soft variable knee design transfer curve where ratio increases over input gain (respec-tively DRIVE) until infinity in a stepless fashion.
On top of that the attack times in LIM mode are ten times faster by default.
Both modes are roughly matched output level wise. Depending on audio material and actual gain reduction applied, this needs to be manually adjusted.

Dynamic range control

Density mkII features a specific control to limit the actual applied amount of gain reduction. Turning RANGE to right-most position leaves the gain reduction unaltered. Tuning it to 12-o-clock position limits GR to about -12dB maximum and leftmost is (almost) zero. One can check the effect instantly with the GR meter to the right.

Note: If there is no GR happening below lets say -12dB then the RANGE knob (obviously) does not have any effect until it's dialed in clearly to be on the right side of the middle mark.

Note: If heavy gain reduction occurs constantly over time and the dynamic range is limited overall below that, then this typically imply that there is no gain riding going to occur anymore. The audio signal is just lowered to a constant level by the compressor (which is probably not wanted). Raise the RANGE control then or lower DRIVE until actual gain riding occurs.

Attack and release time behaviour

The TIMING parameter provides different fixed attack/release time combinations as follows (all values are specified in milliseconds):

	Attack (COMP)	Attack (LIM)	Release
P1	2	0.2	300
P2	2	0.2	800
P3	4	0.4	1000
P4	4	0.4	2000
P5	5	0.5	3000
P6	8	0.8	5000

However the attack times can be relaxed by dialing in the STRICT-RELAX screw between the TIMING knobs turning it from left to right. This then introduces program-dependent timing adjustments on both channels. This concept is basically introduced to achieve more flexibility on the the rather short default attack timings and thus can preserve the audio materials punch in some bus compression situations.

Using this to your advantage:
If there is more peak catching performance needed in your mixing situation then set the dial to STRICT. This is cool e. g. to achieve a consistent dynamic response on difficult audio material such as recordings of strummed string guitars, acoustic bass or so, and this works not only but especially well when the compressor operation is set to LIM mode.
If your audio needs more punch or if too much distortion is already introduced by compression (might be observed as bass implosion or unpleasant audible distortion on some sensitive material) then just relax the timing constants by turning the dial to RELAX.

Why is there a fixed TIMING control but not some traditional ATT/REL behaviour? There are several reasons for that. The first is "simplicity" in the overall operation but this is getting much more important while working in M/S mode. Adjusting time parameters in M/S mode is pretty much tricky and is more easily done with just one TIMING parameter per channel. The overall strict or relaxed timing behaviour is then simply adjusted with the (global) STRICT-RELAX dial which applies now to both channels. As an added sugar, this appears to be a sometimes excellent target in respect to host automation: If you are in the need to adjust and treat the actual tim- ing behaviour differently over a whole track it's now sufficient and very convenient in most cases to just automate the STRICT-RELAX parameter (and not a bunch of channel individual parameters)! Try this trick for sophisticated automated gain-riding over a whole track e. g. for (group) vocal tracks (or anything else with difficult dynamic levels)!

Dual channel processing modes

Density mkII offers three different modes on how the two channels processing is overall managed:

- L/R – the device behaves as a dual mono device where the knobs for each channel can be oper- ated independently. The upper channel is the left signal and the bottom channel is the right one.
- STEREO – normal stereo operation where all (channel specific) knobs are linked.
- M/S – the device behaves as a dual mono device where the knobs for each channel can be operated independently but internally the audio is transcoded into a mid/side signal. This way the upper channel maintains the mid signal now and the lower channel operates the side signal. After compression the signal is decoded back into a standard stereo signal.

Note: Please note that selecting the channel processing mode is independent from internal audio signal linking in the side-chain path (see next chapter)!

Side-chain processing

Linking

This is done with the LINK screw below the RANGE knob and always computes the actual side-chain signal for each channel
in the following way (independent of any compression or channel operation mode):

- In 0% position both channels are unlinked and so each channels side-chain input triggers just the very same channel and not the other.
- In 100% position both channels are linked by adding their side-chain signal (and thus taking the mid part aka the mono content) and triggers now both channel compressors the same way.
- Both positions can be blended in a seamless / stepless fashion.

Tip: *When starting with M/S compression make sure to start with 100% linked mode since this is easier to control and to start with. In advanced M/S compression situations set linking to 0%.*

Filtering

Specially designed for ultra smooth bus compression Density mkII features a fixed shaped filter for filtering the side-chain signal. The filter shape resembles the design from some classics and slightly reduces low frequency content while gently featuring the HF. It can be activated with a switch to the bottom-right. In some cases this might not fit the actual mixing situation where a different filter design is needed – use the external side-chain input in this case and perform your very own and individual filter treatments upfront the side-chain inputs.

Tip: *Use this filter to further avoid pumping effects during compression.*

External

Switching the side-chain input from INT to EXT selects the Plug-Ins audio channels 3 and 4 as the source to obtain the input to calculate the com- pression information.

Important: Make sure that there is an appropriate audio feed into those channels when switching to external side-chain otherwise no compression is going to happen at all!

Tip: *Use the side-chain routing from the kick-drum or bass audio source to achieve those desired pumping effects. De-activate the internal side-chain filter then.*

M/S compression handling

If you would love to have a simpler way to sophisticated mid/side compression here are some tips:

- Make sure LINK is set to 100% first.
- Start setting up the compressor in STEREO mode and dial in what you nor- mally would do w/o having a M/S option on board.
- Now switch to M/S.
- Slightly lower the LINK screw (to around 3-o-clock position or so) to obtain some nicely enliven sound (true stereo input signal required).
- Alternatively to or in combination with the last step try to increase the DRIVE on the side signal and compensate the level drop with the sides MAKEUP dial, so the side signal gets more compression. Use this for easy and consistent stereo width enhancements.

Tip*: If you would like to dig into more sophisticated mid/side compression techniques make sure to unlink (LINK 0%) both channels so the side-chain triggering becomes channel independent!*

Example: Depending on the mix situation it might be necessary or desired to have different compression timings available on the mid opposed to the side channel compressor. This leads sometimes to an inconsistent overall loudness performance between both channels or to too much asymmetric dynamic response. In Density mkII this can easily be overcome by the dynamic range control which is provided by the RANGE knobs and is available per channel. Just limit the actual applied gain reductions for both channels to a similar or appropriate amount so that the resulting compression behaviour appears more consistent between both channels.

Important: To take advantage of this dynamic range limiting technique in M/S mode it is necessary to set LINK to 0% (or nearby) since otherwise just the RANGE setting from the mid channel would be taken into account for both channels.

Dry/Wet mixing

Dry/Wet mixing is a common technique mostly used when the signal is really heavy compressed but then afterwards the uncompressed (dry) signal gets mixed back in. In Density mkII this is achieved just by adjusting the DRY/WET dial and while doing so everything remains phase consistent.

Addendum

Known issues

The Density mkII audio Plug-In is currently not compatible with older SSE (SSE1) type CPU's. There is at least SSE2 required.

Troubleshooting

WTF, there is absolutely no compression going to happen on the compressors output signal? So what to do now? Don't panic, be cool and just check (in the following order):

- Is the compressors power light on?
- Check the “DRY/WET” mixer screw and turn it clockwise.
- Check if SC input is set to INT. When set to EXT make sure that there is significant audio actually feed to the external side-chain input of the compressor.
- Check the RANGE knobs: Turn them clockwise (to the right) so that the GR meter shows actual gain reduction.
- Check the DRIVE knobs: Turn them clockwise (to the right) so that the GR meter shows actual gain reduction.

Rescue & Rescue AE

OVERVIEW

'Rescue' - analog style modelled signal designer at a glance:

- introduces spatial imaging as well as sonic resolution improvements to program material
- simple yet powerful solo signal alteration: transient shaping, imaging and depth balancing
- subtle analog style signal colouration
- easy emphasizing of signal attack phases
- creates "in the face" sounds as well as subtle stereo enhancements
- stereo field operation maintains mono compatibility
- signal saturation at high peak levels
- excellent for drum group improvements
- tightens lowend

features and tech notes:

- minimum latency processing
- low cpu usage
- stereo widening preserves mono compatibility
- inherent mid/side processing
- gain and limit operates on per channel base
- switchable signal colouration
- dynamic noise model
- state of the art signal processing and modelling algorithms implemented
- plugin integration is done with Synthmaker software
- performance crucial parts are written in assembler
- 'ANALOG' section is SSE optimized (the m/s section can't)

BASIC OPERATION AND ADVICE

Use this plugin as an insert effect in any stereo or mono channel of your VST host. Be aware that some of the sonic effects introduced by Rescue just work on stereo program material. However, it can be operated both as a mono or as a stereo plugin.

Tip: *If you are not familiar at all with mid/side processing please refer to audio engineering resources and get used to it before using this plugin.*

Assure the (yellow) On/Off switches to be in 'On' position. Dial in small amounts of the 'WIDTH' knob to apply subtle stereo widening to your audio. A mono signal or the mid audio information of a stereo signal remains untouched.

Dial in the 'PUNCH' knob to apply more punchyness to a mono signal or the mid section of a stereo signal. Alternatively use the 'GAIN' knobs in the 'MID' or 'SIDE' section to perform basic mid/side altering.

Use both to alter the audio signal to taste but remember: a little goes a long way - in most mix situations you will obtain better results by focusing on either improving the punch OR improving the stereo information of that channel. Use the 'VOLUME' knob to adjust the overall output volume as needed.

ADVANCED USAGE

Turning the 'POWER' switch (left side) to 'OFF' disables the just applied effects but preserves the overall sonic 'fingerprint' of this plugin: Like real analog gear this plugin introduces subtle signal alteration just by inserting it right into the signal chain, e.g. frequency and phase response, noise, crosstalk et al. Further non-linearities or signal dependent alterations appear due to the dynamic processing in the mid/side sections. Depending on your monitoring situation and listening experience you can probably identify some of this rather subtle effects by carefully A/B testing at equal RMS volume levels.

Tip: *Use the 'VOLUME' knob on the right side to adjust the overall output and use your VST hosts bypass switch for convenient A/B testing.*

Unlike other "analog" simulations this plugin takes advantage of a dynamic noise model. You won't see any noise at the output of the plugin as long as no input occurs. Instead it will be dynamically introduced. If many Rescue instances are used in a whole mix this assures that the overall noise level doesn't mess up in an unpleasant way (especially in low volume sections). However, if you don't want this (and the other non-linear effects) to appear in your mix at all you can simply turn it off by switching the 'ANALOG' switch to 'Off' position (yellow light of the button dims out). In addition this saves around half the (CPU) processing power needed for the whole plugin.

Assure the 'POWER' switch on the leftmost side is now set to 'ON' to perform the following mid/side processing.

Use the yellow 'MID' and 'SIDE' buttons to activate or disable the specific channel. If both are switched off no audio (and no noise) leaves the plugin. Use this switches for further signal judging purposes while adjusting the both channels as well. This way you can easily identify certain impacts applied to mid or side of the overall (stereo) signal or obtain better decisions on certain applied effect levels. Both switches do *not* work in plugin's OFF mode ('POWER' switch is OFF).

The basic signal flow after mid/side encoding is:

PUNCH -> GAIN -> LIMIT -> Cliping Indicator (for the MID section) respectively
WIDTH -> GAIN -> LIMIT -> Cliping Indicator (for the SIDE section).

Afterwards mid/side decoding applies and the signal is feed into the volume stage and output.

Given that, you are now probably able to perform some different treatments on both channels. Start for example with pretty simple gain adjustments on both channels using the 'GAIN' knobs in each section to alter the signals center or side information. Use the special 'PUNCH' and 'WIDTH' knobs to alter the gain structure just on the signal transients. Combine both as needed.

Important: If there are no transients in the audio content then no effect will happen.

On certain rhythmic and complex program material there will appear a kind of pumping effect when using 'WIDTH' or 'PUNCH' to much or in heavy combination - if so just lower the amount of 'WIDTH' or 'PUNCH' if pumping is unwanted, use limiting (see below) or try to fine tune the applied effect by using the 'RAMP' and 'DEPTH' controls right between both sections. They particularly shape the attack and release behaviour of the transient detector unit. Both are rather more subtle effects and in most situations you should leave 'RAMP' and 'DEPTH' in default position. However, lower 'RAMP' values typically performs better on acoustic material while higher values perform better on electronic. 'DEPTH' can be used to (subtly) improve depth information on certain stereo material.

Tip: *Lower 'RAMP' values typically performs better on acoustic material while higher values perform better on electronic material.*

Tip: *Use the limit function in each section for further signal alteration by turning the 'LIMIT' knobs clockwise.*

In position '0' the limiter transfer function is exactly linear and the signal passes untouched. Turning knobs clockwise introduces and now performs more and more limiting. But: depending on the level of the plugins input signal you probably won't notice any clipping and limiting at all. Use your ears for judgement and also the clipping LED's left and right beside the knobs which indicates if clipping would (normally) happen.

Note: The plugins clipping LED's are just rough and not precise indicators.

Note: Due to the plugin's 32 bit internal floating point resolution actually no real clipping occurs *inside* the plugin but after leaving the plugin's output clipping may occur in your plugin chain or host software.

So, this is mostly a further artistic or "balancing" effect (separate on each channel) and smoothes the gain structure of the overall signal for easy handling outside the plugin. Note also that the clipping indication on mid/side is a completely different thing than L/R clipping outside the plug. You may notice as well that you probably won't see any 'SIDE' clipping on regular recordings unless the side section is heavily driven. This is because in typical recordings most of the audio signals energy resides in the center (mid) and not in the side signal. Be aware that additional harmonic distortion is introduced to the signal by driving a signal hard into the limiter. This may be unwanted (e.g. on clean acoustic recordings like piano, acc. guitar etc.) or may be wanted as an special and appropriate artistic effect (e.g. on electronic instruments or for kickdrum shaping).

Finally adjust the output volume to your (and your host and plugin chain) needs using the 'VOLUME' knob.

EXAMPLES

A true acoustic stereo drumloop recording is already optimized by Rescue while using the 'PUNCH' parameter. A rather low 'RAMP' value supports the natural appearance of this effect. Due to this increased punch in the signals center the overall signal impression appears more upfront (the desired effect in this case) and opposed to that the signals room information is decreased due to the attenuation of signals side information. Compensate this (if ever wanted) by applying a little 'WIDTH' as well as 'DEPTH' or a little 'GAIN' in the width section.

Tip: *Hit the limiter on drum group tracks to obtain sonic grip.*

A stereo group channel needs basic improvement concerning stereo width: Raise 'WIDTH' up to '1' (depending on the transient structure of the material) and increase 'GAIN' in the opposite (mid) section just a little over the top and compensate exactly this with the limiter - so you'll additionally get a little more compact center right as you go (without any compressor usage).

Tip: *Experiment with "the big knob" in one section while twiddling "the small knobs" in the other section.*

Check out some presets as well to get some further inspiration of how to use Rescue as an creative effect.
However, this plugin was mostly designed to obtain subtle and smooth sound enhancements in the digital domain.

Tip: *If you combine 'GAIN'/'PUNCH' or 'GAIN'/'WIDTH' then just lower or improve the amount of audible transient amount by adjusting the 'RAMP' knob.*

Rescue AE – The Blackface Anniversary Edition

Due to the 1.2 release this package now introduces the additional 'RescueAE' Plug-In.

That new Plug-In features a slightly different signal path, some internal rework, 4x Oversampling and a brand new output limiter circuit.

The output limiter is placed right behind the output 'VOLUME' knob so you can additionally drive or reduce the amount of limiting to occur in the output stage. This is completely new and not available in the normal version.
Both, limiter and 'VOLUME' knob, are disabled when switching 'POWER' off. This way the 'RescueAE' versions signal path behaves slightly different compared to 'Rescue' where the 'VOLUME' knob stays always active even when powering off the Plug-In.

The release parameter of the limiter is fixed to 300ms and there is just one option available: Engage or not. However, the release parameter is available internally, so if your host supports GUI-less operation of a VST fx or automation then you can still access the release time of the limiter (ranging then from 10ms up to 1sec).

The (soft-) limiters in the 'MID/SIDE' section of the AE version are now 4-times oversampled for a much smoother audio experience even on acoustic material.
The oversampling filters are optimized for minimum latency and CPU usage but therefore are not linear in phase. So please keep in mind that this is a coloring device and does not act transparent to the sound.

And now some last tips ... FURTHER TIPS & TRICKS

- Level your audio input to the plugin to around 0dbFS to perform easy and best inside the plug.
- Use <ctrl> + click on a knob or switch to restore default position.
- Use <shift> + click on a knob to fine adjust values.
- Some Knobs snaps gently into default position if moved slowly over it!!
- If L/R positioning information behaves strange in your stereo recording it might be better/necessary to fix this in front of any mid/side operations in general.
- If really heavy signal alteration is actually necessary on your material it might be better/necessary to first improve the overall signal chain quality (e.g. in a lower quality recording situation).
- If you notice a drop in volume of lower frequencies while inserting Rescue then your signal is not mono compatible in the lower frequency range.
- You can use Rescue as a DC filter just by inserting ('ANALOG' must be on).
- And always remember: garbage in, garbage out ;-)

Tessla SE

Assure the output switch in the lower right corner is in 'on' position (red lightning). If it's grey click on it until it's red lightning. This toggles the overall plug-in operation (on/off).

Level your incoming audio so the needle of the VU style metering in the middle of the plug-in hits clearly the right most (red) area. Dial in amounts of the 'saturate' knob to apply further saturation effect to your audio if wanted.

OVERVIEW

'TesslaSE' – modelling pleasant sounding 'electric effects' coming from transformer coupled tube circuits in a digital controlled fashion.

at a glance:

- transformer style signal saturation
- smooth transient polishing and increased RMS leveling
- subtle analog style signal colouration
- switchable sub bass enhancements

features and tech notes:

- zero latency processing
- low CPU usage
- harmonic enhancements without artefacts (aliasing)
- proprietary signal processing and modelling algorithms implemented
- plug-in integration is done with Synthmaker software
- performance crucial parts are written in assembler
- completely SSE optimized

BASIC OPERATION AND ADVICE

Use this plug-in as an insert effect in any stereo or mono channel of your VST host. It can be operated both as a mono or as a stereo plug-in.

Use the 'output' knob to adjust the overall output volume as needed and for handy A/B testing at equal volume levels.

FINDING THE SWEET SPOT

TesslaSE performances best as a subtle audio effect applied here and there in a whole mix situation. Don't expect all your mixing problems to be solved by just some magic on the stereo bus – there is no such 'holy grail' in audio processing.

Turn up your selected audio channels volume that way it basically performs around 0dBFS. The 'VU' type meter in the plug-in is adjusted to support you that way: Increase volume so the needle hits clearly the red marked section of the meter.

Increase 'saturate' as needed until the unit performs audible distortion. If you already notice distortion then lower slightly the input volume to the plug-in or turn down the 'saturate' knob counter clock wise – you are now in the sweet spot.

Tip: *Use this type of setting to perform TesslaSE on several audio channels in your whole mix to improve slightly the overall sonic image and density.*

Tip: *Use the 'input' knob to level up the incoming audio volume.*

ADVANCED USAGE / GAIN COLOURATION

Like real analog gear this plug-in introduces subtle signal alteration just by inserting it right into the signal chain, e.g. frequency and phase response as well as a harmonic fingerprint.

Depending on your monitoring situation and listening experience you can probably identify some of this rather subtle effects by carefully A/B testing at equal RMS volume levels.

Tip: *Use the 'output' knob on the right side to adjust the overall output and use the 'on/off' output switch for convenient A/B testing.*

To take full advantage of the transformer style gain colouration provided by TesslaSE please use the switches on the right side of the VU meter. Switching from 'OFF' to +3, +6 or +9dB mode features further specific colouration which usually appear as if you would 'push' real analog gear into its limits. Re-adjust input or output volume if necessary.

Note: The former experimental 'AUTO' mode (as in Version 1.0) has been removed in the actual version.

THE 'phat' MODE

Since Version 1.1 a new 'phat' option has been introduced. This option increases slightly the density of the harmonic spectrum produced by the saturation of the device. Use this feature to taste.

SUBBASS ENHANCEMENTS (SBE)

The core algorithm of 'TesslaSE' performs a sweet bass enhancement which is swichable due to the 'SBE' switch. In the 'TesslaSE' version it's tuned to a fixed frequency around 55Hz and it increases the lowend perception unlike any ordinary EQ.

Tip: *Use the plug-in's default setting to perform the plug-in in a couple of instances on a whole mix to increase the overall sonic impression.*

This may increase the perception of overall depth but density as well.

TesslaSE AS AN ARTISTIC EFFECT

Don't hesitate to use TesslaSE as an artistic effect while driven heavy on appropriate tracks. Good examples of application might be drum group smashing, saturating bass tracks or smoothing digital synth lines.

Tip: *Use the 'input' knob to level up the incoming audio volume until the VU meter needle hits hard the red side.*

Enjoy finest analog style saturation!

CPU USAGE / SAVINGS

The different selectable features of TesslaSE increases the overall CPU consumption of the plug-in. The other way around disabling them will save CPU cycles. The basic 'saturate' feature is always in the signal path and therefore always needs some CPU.

Tip: *If CPU usage is an issue in your mixing situation avoid the colouration feature. 'SBE' and 'phat' just use a little more CPU.*

FURTHER TIPS & TRICKS

- Level your audio input to the plug-in to around 0dbFS to perform easy and best inside the plug.
- Use <ctrl> + click on a knob or switch to restore default position.
- Use <shift> + click on a knob to fine adjust values.
- And always remember: garbage in, garbage out ;-)

Tessla PRO

Usage tips:

- Use the 'OUT' knob to level the outgoing audio and for handy A/B comparisons
- Level your audio input to the plugin to around 0dbFS to perform easy and best inside the plug
- Use <ctrl> + mouse left click on a knob or switch to restore default position
- Use <shift> + mouse left click on a knob to fine adjust values
- Use this plugin as an insert effect in any mono or stereo channel of your VST host

Overview

'TesslaPRO' – transient aware signal saturator.

At a glance:

- transformer style signal saturation
- smooth transient polishing and increased RMS leveling
- transient aware signal saturation
- switchable subbass enhancements
- subtle and nice audio coloration enhancements

Plug-in specification:

- PC / VST compatible
- SSE and Assembler optimized sound engine
- state-of-the-art digital signal processing
- minimum latency processing
- harmonic enhancements with low artefacts (aliasing)
- plugin integration is done with Synthmaker software
- performance crucial parts are written in assembler
- completely SSE optimized

Quick reference

From left to right:

BASS – increase or decrease the bass response
DRIVE – alters the gain amount of the signal into the processor
BOOST – adds further 12dB gain
TRANSIENTS – turning clockwise lets transients pass through
POWER – plug-in on/off
OUT – output level

Basic operation and advice

Use this plugin as an insert effect in any stereo or mono channel of your VST host. It can be operated both as a mono or as a stereo plugin.

Assure the POWER switch is in 'on' position (red lightning). If it's grey click on it until it's red lightning. This toggles the overall plugin operation (on/off).

Level your incoming audio so the grey needle of the VU style metering in the middle of the plugin hits clearly the right most (red) area. Dial in amounts of the 'DRIVE' knob to apply further saturation effect to your audio if wanted.

Use the 'OUT' knob to adjust the overall output volume as needed and for handy A/B testing at equal volume levels.

Finding the 'sweet spot'

TesslaPRO performances best as a subtle audio effect applied here and there in a whole mix situation. Don't expect all your mixing problems to be solved by just some magic on the stereo bus – there is no such 'holy grail' in audio processing.

Turn up your selected audio channels volume that way it basically performs around 0dBFS. The 'VU' type meter in the plugin is adjusted to support you that way: Increase volume so the needle hits clearly the red marked section of the meter.

Increase DRIVE as needed until the unit performs audible distortion. If you already notice distortion then lower slightly the DRIVE – the plug-in is now in it's sweet spot.

Tip: *Use this type of setting to perform TesslaPRO on several audio channels in your whole mix to improve slightly the overall sonic image and density.*

Tip: *Use the BOOST switch to level up the incoming audio volume if needed.*

Advanced usage / Transient processing

Like real analog gear this plugin introduces subtle signal alteration just by inserting it right into the signal chain, e.g. frequency and phase response as well as a harmonic fingerprint.

Depending on your monitoring situation and listening experience you can probably identify some of this rather subtle effects by carefully A/B testing at equal RMS volume levels.

Tip: *Use the OUT knob on the right side to adjust the overall output and use the 'on/off' output switch for convenient A/B testing.*

To take full advantage of this plug-in you can use the TRANSIENT dial to alter the saturation processing on the audio signal transient information.

Turning this knob straight to the left means that all signal information gets processed. Turning the knob clockwise then lets more and more transient information passing (unaltered) by the saturator.

Tip: *The red 'VU' meter needle indicates if there is transient information passing through. If all transients gets saturated the needle stays to the left. This is also the case if there is no transient information available at all in the actual audio signal.*

Subbass enhancements

The core algorithm of 'TesslaPRO' performs a sweet bass enhancement which is adjustable due to the BASS dial. It does not only perform equalizing but saturation and dynamic processing as well.

TesslaPRO as an artistic effect

Don't hesitate to use TesslaPRO as an artistic effect while driven heavy on appropriate tracks. Good examples of such applications might be drum group smashing, saturating bass tracks or smoothing digital synth lines.

Enjoy finest analog style signal saturation!

FerricTDS

Usage tips:

● Use the TRIM knob to level the outgoing audio and for handy A/B comparisons
● For optimum processing and results, the input level to the plug-in could be adjusted to peak around 0dBFS
● Alternatively, use the INPUT adjustment knob to bring the incoming signal onto duty level (plug-in internally)
● Use <ctrl> + mouse left click on a knob or switch to restore default position
● Use <shift> + mouse left click on a knob to fine adjust values
● Use this plug-in as an insert effect in any mono or stereo channel of your VST host

Overview

FerricTDS – Tape Dynamics Simulator.

Inspired by the smooth dynamic shaping capabilities of some high-end reel-to-reel tape recorders, this plug-in simulates three of the most distinctive and much appreciated sonic effects generated by these devices:

● DYNAMICS – gently shaping the overall dynamic response
● SATURATION – adding extra harmonic-related content
● LIMITING – controlling peak performance

Other rather ugly side effects, such as tremendous phase and frequency alterations, wow and flutter, noise and crosstalk and others, are not included in this simulation.

Functions at a glance

● performs gentle audio dynamic treatments
● masters difficult to handle audio material
● adds extra harmonics and saturation effects
● controls outgoing audio peaks

Plug-in specification

● Win32 / VST compatibles state-of-the-art digital signal processing
● performance-critical parts are written in assembler
● completely SSE optimized

Getting the most out of it

Please read the following chapters to get the most out of this device. Learn how to efficiently set the three main parameters:

● obtain some effective tips on getting the most out of the presets
● understand the basic workflow of this device
● learn how to efficiently level the three main parameters
● take advantage of the RECOVERY and sidechain options in the DYNAMICS section

Quick reference

Knobs and “screws”:

DYNAMICS – increases audio compression type effects
SATURATION – increases saturation type effects such as harmonic additions
LIMITER – controls the overall peak performance
RECOVERY – controls the recovery time of the dynamic processor
TRIM – adjusts the outgoing audio level
SC OFF / 250Hz – this “small screw” controls the dynamic processor's response to low frequency material by a high-pass filter
INPUT – adjusts the incoming audio level

Switches:

ON / BYPASS – basic on/off operation. The VU meter still operates in bypass mode to allow the user to set optimum input level
MOD/CLASS – the tape mode switch which selects between a “classic” and “modern” sound processing

Meters:

Main “VU” meter – indicates the outgoing signal level (averaged)
Horizontal meters – roughly indicate the amount of signal processing in the DYNAMICS (left) and the SATURATION (right) processors

Basic operation and advice

Use this plug-in as an insert effect in any stereo or mono channel of your VST host. It can be operated both as a mono or stereo plug-in.

The best performance is obtained if the audio input is leveled to around 0 dBFS peak performance. In BYPASS mode the VU style meter needle should occasionally hit the red marked range. This indicates the “sweet spot” of the device. Alternatively, the INPUT dial could be used to compensate the incoming audio level instead.

Make sure that the BYPASS switch is in ON position now. The two horizontal metering displays on the top left and right side are now responding to the incoming audio if processing actually occurs.

Next, dial in some DYNAMICS to obtain compression effects and SATURATION for some gentle distortion effects. Depending on the audio material, this effect can be rather subtle. Increase the input level to the plug-in if the effects appear too subtle.

Use a combination of both effects, then add some limiting by turning the LIMITER knob clockwise to block outgoing audio signal peaks. Since version 1.5 this works as a true and accurate brickwall limiter.
Use the TRIM knob to adjust the overall output volume as needed. This feature is also handy for A/B testing at equal volume levels.

Some tips on using the presets

Explore the presets, but always adjust them to your current mixing situation:

● Always adjust the DYNAMICS and SATURATION parameters to your current track or mix in order to adapt the actual processing to your specific needs.
● Always adjust the TRIM parameter to your current mix for equal loudness lev- els during A/B comparisons.
● All presets were designed on audio material with levels peaking at around 0 dBFS.
● Take advantage of the small INPUT dial to level up or down your incoming audio to fit to a preset setting.

Advanced Usage

Internals

Different waveshaper types

Under the hood, FerricTDS consists of quite a bunch of different waveshapers and envelope generators. One of them, for example, is used to obtain a frequency sensitive gain reduction signal to be used by the DYNAMICS processor, which works hand in hand with an additional processor that generates odd order saturation effects (con- trolled by SATURATION). Finally, peak control is performed by yet another processor.

You can regard this as one single circuit with just some different parameter controls. However, there is one exception to this: Straight peak limiting occurs not till the output of the device, after all other processing has already taken place. It is one important consequence of this design that all peak and transient information which passes the prior stages has to be handled in the output limiting stage. Otherwise, it will remain unmanaged when leaving the device. This design leaves it to the user to decide how much peak and transient information actually leaves the device. It also permits the use of an external limiter of your own choice if desired.

The limiter reacts instantaneously to incoming peaks, offers an analog style transfer curve (similar as shown in the diagram above) and is a true and accurate brickwall limiter. Due to this design, both aspects are regarded: The tribute to the “tape” concept where no hard-knee thresholds can occur and the same time “zero overshot” performance is guaranteed as well.

Given this short introduction, two different workflow approaches are recommended, depending on your specific goals when using FerricTDS:

1. Precise dynamics handling or
2. limiting and maximizing.

In both workflow scenarios 1 or 2, always start tweaking with DYNAMICS, SATURATION, and LIMITER set to the leftmost position and SC set to OFF (as in the default preset).

Workflow 1: Starting with DYNAMICS

To get the dynamics most accurately handled by FerricTDS, start by adjusting the DYNAMICS knob until you achieve a suitable amount of compression. Fine-tune the processing with the RECOVERY option. For example, slow the response time if too much compression is applied to snare drum hits in a mix-bus or mastering application.

If necessary, use the SC parameter to adjust the compressor's response to bass. Now, dial in SATURATION to obtain a blend of saturation effects. In the last step, dial in LIMITER amounts if more peak control is necessary or wanted.

Workflow 2: Starting with SATURATION

If, in contrast, you are after more saturation and peak limiting control (for more distorted effects or maximizing purposes), start by setting the SATURATION knob to the right. Apply further LIMITER amounts to obtain the level of peak control needed in your situation (or let an external limiter do this job).

As a final step, dial in small amounts of DYNAMICS, but this time with RECOVERY set to 'fast' and SC set to OFF (0 Hz). This enables some smooth overall gain-riding that slightly relaxes the peak limiting module. If pumping occurs, adjust the sidechain high-pass filter or lower the DYNAMICS amount.

The RECOVERY parameter

This parameter lets you control how fast the DYNAMICS processor is recovering from its duty cycle. 'Fast' operation (knob in leftmost position) enables faster gain riding, but bear in mind that a faster operation might introduce more unwanted distortion resulting from audio inter-modulation.

'Slow' settings usually result in more audible compression type effects. This can be unwanted in some cases, e.g.:

- the snare drum in a mix is affected too much by compression
- overall limiting/maximizing is desired

The SC (sidechain) filter option

The SC filter option (the “small screw”) controls a 12dB/oct Butterworth high-pass fil- ter in the sidechain path (not in the audio path itself). It operates from 0 Hz (aka OFF) up to 250 Hz. This filter lets you control the amount of low-frequency audio con- tent that will affect the DYNAMICS processor. This parameter is available to host automation (named “HP”).

Some example HP filter plots

This can be used, for example, to avoid “pumping” compression effects. Be aware that if the LIMITER option is enabled, it must handle major parts of the low frequency content that passes through the DYNAMICS processor untouched.
For further orientation: In the screw's 12-o-clock position, the filter is tuned to 100Hz (at -3dB reference).

The two tape modes

Since version 1.5 an additional tape saturation mode is offered. Both modes are selectable with the MOD/CLASS switch:
CLASS – (similar as offered in version 1.0.2), a type of classic tape saturation which attenuates the bass response and brings in more mid frequency information. HF frequencies might also appear to be more “tamed”.
MOD – a rather modern tape variation which offers a more relaxed sound especially in the bass range and allows more peak control at same output RMS levels.

Limiting and Maximizing

When using the LIMITER at larger input levels you might notice that the overall output volume appear quieter opposed to the unprocessed signal. This is normal with FerricTDS due to the internal gain staging.

However, if you are after audio level maximizing then compensate this effect by simply increasing the INPUT level dial until equal (or higher) perceived loudness is achieved at lowered peak performance.

When using FerricTDS as a maximizer just use the default preset and dial in the LIMITER to 100%. Now drive the unit by the INPUT dial. More sophisticated and program dependent behavior can of course be obtained by utilizing different DYNAMICS and SATURATION settings. For this purpose in most cases the modern tape setting per- forms way better opposed to the classic one.

Dry/Wet mixing

Since version 1.5 FerricTDS has a 100% flat frequency and phase response. This allows external dry/wet mixing of the entire effect which means that one could mix the processed signal back into the unprocessed (e.g. by utilizing send effect configurations in a host).

Non-linearities and level changes

Most parameters are roughly volume compensated, but due to the complex internal design and non-linear behavior, there is no guarantee that accurate compensation can be achieved for all kinds of input sources.

About non-linearities and harmonic spectrum alterations: The main saturation circuit mainly produces odd order harmonics, similar as shown below.

Note: Actual spectrum measurements vary with different parameter settings. The spectrum varies between the two tape modes.

Addendum

A brief history of tape

The concept of magnetic recording to a moving tape was invented by the German-Austrian engineer Fritz Pfleumer and received a patent back in 1928. The basic idea was to translate the voltage from the audio signal straight into magnetic energy, which then induces magnetic particles on a tape (moving along the inductor at constant speed). These particles manage to store the audio information. The whole process goes the other way around for recall.

Although this was a revolution for both broadcast and recording industry, there were many technical challenges to be addressed before its success during the middle of the last century. Some physical limitations can't be ignored even today. While electromechanical problems, such as wow and flutter or noise and crosstalk have been improved over the years, the electromagnetic phenomena, such as magnetic permeability, hysteresis or the Barkhausen effect still must be addressed.

Additionally, since a tape can't store unlimited amounts of energy, a natural saturation occurs when signal levels are driven too hot. Normally this has to be avoided, as it can lead to heavy distortion. Nonetheless, this type of saturation was (and still is) frequently used as an artistic audio effect.

The new digital recording technologies that emerged towards the end of the 20th century overcame these shortcomings of analog recording and made tape obsolete – if regarded from a purely technical and workflow-related point of view. Yet some of the positive effects of high quality tape and recorders are still highly appreciated in today's audio production, and there is quite a lot of myth and buzz going on about it's “magical” qualities.

In fact, what makes a good tape and recorder still attractive in the digital age is its overall ability to balance audio dynamics while adding harmonic content and gently limiting the peaks. If properly applied, this can result in a very pleasant sonic experience. However, it still comes at the expense of some of the mentioned artifacts and side effects, not to mention the time and cost of operation and maintenance.

Judging saturation effects

There are quiet some mistakes floating around on how to judge a saturators sonic quality and here are some tips to avoid the most common pitfalls:

1. A good saturator does not appear as distortion in the very first place. Firstly it just saturates incoming audio signals which means that at a similar RMS out- put level it simply reduces the peak performance (which results in a smaller “crest factor”).
2. This immediately implies that you need a RMS meter in your output chain to compare different saturation settings or devices to another. Basically this is the same when comparing limiters or maximizers.
3. Distortion is a side-effect which typically occurs at higher saturation levels. It can have different sonic qualities, e. g. due to the frequency distribution of distortion which makes a huge difference to human hearing and if the effect is perceived as to be rather gentle or not.
4. Don't rely here on a simple spectrum analyzer since it does not know nothing about the concept of being “gentle” or not.

Summary: Always assure equal RMS output levels and then use your ears.

Nasty DLA

- Win32 / VST compatible
- state-of-the-art digital signal processing
- performance-critical parts are written in assembler
- completely SSE optimized

Usage tips:

- Use the power switch on the right side for handy A/B comparisons
- Use <ctrl> + mouse left click on a knob or switch to restore default position
- Use <shift> + mouse left click on a knob to fine adjust values
- Use this plug-in as an insert effect in any stereo channel of your VST host

Overview

NastyDLA – a classic chorus echo device with tape-delay simulation.

Inspired by the classic analog chorus echo device, this plug-in implements some of the most distinctive and much appreciated sonic effects generated by these devices:

- classic chorus and echo effects
- authentic signal path coloration
- tape-delay style feedback and saturation

Functions at a glance

- applies gentle feedback driven delay effects
- performs smooth audio signal modulations
- shapes frequency and phase response
- adds extra harmonics and saturation effects

Plug-in specification

Quick reference

From left to right according to the graphical interface:

#	GUI	AUTOMATION	DESCRIPTION
0	POWER	POWER	Plug-in ON/OFF operation.
1	INPUT	INPUT	The input stage volume control. With SAT enabled this is a true drive/gain control.
2	SAT	SAT	If SAT is enabled then the input stage changes the signal coloration. Frequency and phase response of the signal are affected and non- linearities are applied.
3	CHORUS	CHORUS	Depth control of the chorus/flanger effect. Left most position means off and is a true bypass.
4	RATE	RATE	Modulation frequency control for this effect.
5	COLOR	COLOR	This is a overall frequency balancing filter (high vs. low frequencies). It's rather subtle but gets more meaning when increasing the amounts of feedback (FB) of the delay section. Mid position is off position.
6	I-II	COLOR-I-II	Switches between a rather flat frequency response (when off) and a more resonant and mid-focused timbre (in on position).
7	FEEDBACK	FEEDBACK	Amount of feedback in the delay circuit.

8	HP	HP	A standard DSP 12dB high-pass filter.
9	LP	LP	The custom "tape-style" low-pass filtering.
10	none	MODE	The big knob right beside the VU meter selects one out of the seven different delay modes.
11	NOISE	NOISE	Adds simulated “tape hiss” noise.
12	FEEL	FEEL	Adds negative or positive pre-delay.
13	DUCK	DUCK	Switches the internal tape compressor into ducking mode.
14	ECHO1	ECHO1	The first echo time control slider.
15	SYNC	SYNC1	
16		SYNC2	Syncs ECHO1 and/or ECHO2 to host.
17	ECHO2	ECHO2	The second echo time control slider.
18	AGED	AGED	Adds further phase smear.
19	MODULATION	MODULATION	Delay time modulation. Left most position means OFF.
20	I-II	MODULATION-I-II	Switches between two internal types of modu- lation.
21	DRY	DRY	Dry signal amount.
22	WET ONLY	WET-ONLY	"Wet Only" control. Suppresses the dry signal.
23	WET	WET	Wet signal amount.

Delay modes explained

#	GUI label	Description
1	mono 1	One single classic "tape-style" echo. Just the ECHO1 controls are in charge in this configuration.
2	mono 1&2	Same with both echos. This is still configured as mono.
3	dual mono	ECHO1 operates on channel one and ECHO2 on channel two.
4	ping pong	The typical ping-pong style echo configuration. The input is taken from channel one and then ECHO1 is applied first and output to channel one and feedback to ECHO2 which outputs to channel two and then again feedback to channel one and so on.
5	cross feedback	This is a dual mono configuration but with both channels cross changed in the feedback path. This configuration gives highest diffusion if feedback is applied.
6	retro 1	Both "retro" modes are changing the routing in general: The DRY output is wired to channel one and the WET output is wired to channel two. Otherwise similar to the "mono echo 1" program.
7	retro 1&2	Same here but with both delay lines activated and controllable.

Advanced

Internal architecture

Internally, NastyDLA consists of quite a bunch of DSP processing building blocks which as a whole are summing up to an authentic signal path simulation of it's analog models. The blocks and the according signal flow are shown in the diagram above. Basic signal flow goes from left to right except the feedback path which goes in the opposite direction.

With NastyDLA, signal path coloration already starts in the input stage which provides a complete model of both, frequency and phase response as well as dynamic saturation. It's located in the dry path but all nonlinear processing and coloring can be disabled (via the SAT switch) on demand so it remains as a simple input volume control then. But while switched in, the input stage can greatly contribute on getting the processed signal to fit right into a mix.

The pre-delay block changes the timing between the following dry and wet signal paths and can be positive or negative which means that the wet signal can be delayed related to the dry path but the opposite as well. This can be dialed in with the FEEL knob in the interface.

The chorus algorithm is hard wired and is the very first block in the wet signal chain. Internally it computes four audio lanes and depending on the operation mode of the plug-in (mono, dual-mono/stereo) they are dynamically configured to compute the appropriate audio output. Rate (RATE) and amount (CHORUS) can be adjusted to obtain a broad effects spectrum ranging from subtle and smooth chorusing up to flanger like effects.
Next in the chain is the delay algorithm itself which offers seven different configurations (two mono modes, dual-mono, ping-pong, a cross feedback delay and two retro modes) and contains additional modulation options. Given the modulations, static audio delay imaging can be avoided and even tape-like timing imbalances can be obtained. This is also the place where a sophisticated tape hiss simulation takes place which basically implements a colored and animated noise model (NOISE). With the two retro modes the dry signal is routed to the left audio output channel and the wet is to the right, being a reminiscent to the analog originals.

Serious amounts of additional phase distortion can be introduced with a dedicated phase section behind the delay block. This can be used for an increased dispersion of re-occurring delays in between the feedback path and contributes to a more “reverb-ish” sound and also the typical screaming sound when the units feedback is driven right into self oscillation. This option is accessible through the AGE switch.

NastyDLA contains a comprehensive and detailed modeled dynamics section. It consists of a compressor/saturator tandem where not only static waveshaping is applied but subtle frequency dependent compression as well. The basic technology for this is taken from the already released and award winning FerricTDS tape dynamics simulator. The compressors sidechain can be switched from the processed feedback path back to the dry signal to provide a slight audio ducking effect (DUCK). Also, the saturator itself is not static at all but aware of dynamics too. Altogether an authentic and consistent saturation experience can be achieved. The compressor and saturator are always in and can't be disabled.

Last in the feedback loop resides EQ and filtering. While the frequency adjustable high-pass is a standard DSP 12dB/octave filter (HP), the low-pass filter (LP) is a custom design which highly contributes to the overall specific sound especially when feedback is used. Most digital tape delay emulations just offer plain standard DSP resonant lowpass filters where increased resonance is utilized to achieve screaming effects. The obtained results sounds harsh and plastic quite often. NastyDLA avoids this and the desired and typical feedback driven sound is created solely based on smooth filtering, dynamic saturation and massive phase distortion in the feedback path. Instead of a conventional EQ, a simple but effective “niveau” leveling filter (COLOR) is implemented which makes it dead easy to change the overall audio tonality. It's center frequency adopts automatically to the actual frequencies of the LP/HP filters.

All that meticulous modeling entirely the way through the whole signal path comes to a price at the end: the computational cost. Nonetheless, decision was made to have a no compromise technical design for NastyDLA available and not just yet another flat and digital sounding delay.

In use – some practical tips

Taking advantage of audio signal coloration

Audio signal coloration takes place in a couple of different circuits in NastyDLA. The most obvious is of course the EQ and filtering section (COLOR, LP, HP) where frequency and it's dependent phase manipulation which are applied in a direct fashion. This is typically the starting point if one chooses to dial in some specific timbre and allows some significant change of the delay tonality if FEEDBACK is used. In the default setting it provides a rather flat frequency and phase response and with the I-II switch engaged it changes to a more resonant and mid-focused sound – again, if FEEDBACK is used. Turn COLOR from mid position to the right to obtain a brighter tone and to the left to obtain a darker tone. As an example, dial in some darker tone with the COLOR option (moved to left) and to retain and not loose the mid focus engage the I-II switch.

Tip: *Always adjust the FEEDBACK level to match the same level of resonance when comparing different modes.*

The AGED option offers an additional and drastic method of manipulating the signals phase and (simply speaking) applies more phase smear and distortion as in older tape media or devices. Use this creatively for more delay line diffusion, for example, or to obtain different delay colors when using shorter FB settings. However, to some this might be rather subtle, depending on the actual delay time and source material.

Tip: *Try it with rather short delay times to hear what it does.*

The input stage colors quiet a lot as well but only if SAT is engaged. Then, a basic device frequency and phase response is applied but more important, non-linear behavior takes place in the whole signal path. This goes way beyond static waveshaping and offers a consistent saturation experience over a wide frequency and input volume range.

Proper embedding in the mix

The aforementioned input stage saturation can greatly be utilized to get your chorus and delay signals to fit right into your mix. A more compact mid frequency range and tighter dynamics can make the final mixing way easier.

Adjusting the pre-delay option (FEEL) is another method and can drastically help to improve the delay line sitting right into it's place comparing to the dry signal. Try posi- tive as well as negative settings to discover what your mix groove actually needs.

Proper EQing is the key, of course. The COLOR (niveau-style) filter is the fastest and most easy way to change the wet signals frequency appearance. Turned clockwise it supports a brighter sound impression and vice versa. This may be subtle if no FB is used but becomes more prominent when FB is dialed in more and more.

Tip: *Enabling the I-II switch beneath the COLOR knob gives a more mid- focused and resonant tone.*

With the HP filter option, unwanted low frequencies can be excluded from the party quite easily. This is done with a 12dB/octave high-pass filter which is a good compromise between smooth and steep filtering. Though, sound wise more important is the dedicated low-pass filter which is capable of shaping and attenuating the high mid/high frequencies in a “tape-style” manner. It offers the proper and smooth “roll-of” so that too much high frequency information (which is not contained in a natural echo) is avoided.

In the dynamics section the ducking option (DUCK) can definitely aid in easier mixing as well. If enabled, the wet signal slightly gets attenuated as soon as an input signal emerges. When the input signal pauses the internal compressor is going to relax and the wet effect signal raises slightly.

Note: Adjust the actual FB settings when using the ducking mode since feedback handling and self oscillation behavior differ depending on the compression mode.

Warning: In ducking mode, feedback and self oscillation behavior may appear much louder and blew up your speakers or cans (and ears).

Understanding chorus and delay modes

For the sake of simplicity, in chapter 3.2 “internal architecture” the channel specific routing is omitted for all the different delay modes. These modes partially also affects the chorus. To fully understand the internal routing and modes just a few principles are necessary:

● The plug-ins outputs is audio always on two channels (thus, a stereo channel is necessary to insert this plug-in).
● The mono programs are taking the audio input just from the first (left) audio channel (“mono 1”, “mono 1&2”, “retro 1”, “retro 1&2”) and the “ping-pong” program does so as well.

Note: In this modes the chorus is a true 4x chorus.

● “dual mono” and “cross feedback” takes input on both input channels.
● The “retro” modes splits the dry signal to the channel 1 output (left) and the wet signal to channel 2 output (right).

The chorus is changing it's configuration depending on the selected delay mode. In mono delay configurations it works internally as a 4x chorus and in a dual channel or true stereo setup it works in a 2x2 configuration.

Delay time modulations

There are two delay line modulation options provided with the I-II switch right beneath the MODULATION dial. Engaging this unveils a subtle but more chorus alike sound and this is suitable to improve the overall depth impression of the computed echo's. Combine this also with the CHORUS itself for maximum diffusion. If a random but more dry experience is wanted then leaving this switch off is the ticket.

Note: This I-II switch changes the animation of the actual noise (as being dialed in by the NOISE parameter) as well.

If an echo's delay time signature is not synced to host (via the SYNC switches) then both ECHO parameters can be automated via host automation or manually animated to obtain smooth “tape speed” changes.

Compression and saturation

There is always a tape compression sort of thing working under the hood which handles both, compression as well as dynamic saturation aspects. It can be switched into a ducking mode where its sidechain path gets routed to the plug-ins dry path and given that the compression becomes dependent to the volume of the actual input sig- nal. This provides a slight ducking effect and is activated by the DUCK switch.

Getting the tape delay feel

If you want to obtain a rather tape-style feel with NastyDLA, here are some tips as a basic guidance:

● Select one of the mono or retro modes.
● Set the LP filter somewhere between 1k and its mid position (ignore the HP switch or set it somewhere nearby 20).
● The COLOR knob (plus its I-II switch) deploys a broad range of different timbres. This way, a lot of different tape delay timbres can already be achieved.
● Its recommended to use the input stage as well (SAT) to dial in some further amounts of non-linearities and crunch.
● Dial in some amounts of MODULATION but turn its I-II option off.
● Always use the NOISE option and don't hesitate to go up to -80dB or an even higher noise-floor.
● The AGED switch applies even more phase smear – thats what old tape media basically does.

Note: Although the original devices does not have a ducking mode it might be useful to turn DUCK on if more tape saturation is desired. In doing so, the compression sidechain is not obtained from a signal in the feedback path anymore and therefore the saturation stage is left to do a little bit more work. Could be great if you are after that screaming sound.

In doing so, NastyDLA rewards you with the possibility of not only simulating just one single model but quite a bunch of different tape delay style timbres instead.

Tip: *If you are sound-wise there then don't hesitate to combine this with more modern options such as a cross-feedback delay routing or take advantage of the ducking feature, the FEEL option or add some gentle chorus and work in true stereo – things no original or replica can give you.*

About the retro modes

Similar to how some vintage models worked to their time, the two retro modes just produce a mono effect signal but the plug-ins output is stereo now: dry is routed to channel one (left) and wet is routed to channel two (right). Its just a reminiscent to that models and if one would love to have that design today in a plug-in its provided now with NastyDLA.

Note: The WET ONLY feature does not work with the retro modes by intention (its just disabled then).

Addendum

The classic chorus echo device

There are just a few audio effects available that are capable of instantly turning a small and wimpy riff into something big and meaningful. One of them is the classic chorus/echo combination. Beside the individual classic echo or chorus devices these combined devices were historically build around true tape or bucket brigade delays.

From today's production standards perspective they might be easily overseen (feature wise) but on the other hand they are still pretty much demanded due to their specific and warm tone and this unique sound quality is probably the charm which still today attracts producers and audio engineers to use them in their actual music productions.

NastyDLA is going to follow this path and recreates all the specific tone qualities while adding just some few but well selected modern features. The plug-in implements some of the most distinctive and much appreciated sonic effects generated by these devices:

- classic chorus and echo effects
- authentic signal path coloration
- tape-delay style feedback and saturation

NastyDLA applies gentle feedback driven delay effects, performs smooth audio signal modulations and adds extra harmonics and saturation effects.

epicVerb

Usage tips:

- Use the 'OUT' knob to level the outgoing audio and for handy A/B comparisions
- Use <ctrl> + mouse left click on a knob or switch to restore default position
- Use <shift> + mouse left click on a knob to fine adjust values

Some general tips on reverberation:

1. Less is more in todays modern music productions
2. In some cases adding just some early reflections is enough and no full reverb tail is necessary – use the epicVerbs 'AMBIENCE' mode for this
3. Feed the audio through a delay into the reverb to achieve that "larger then life" sound
4. Use EQ to remove resonances or shape the overall frequency response of the room simulation

And always remember: garbage in, garbage out ;-)

Overview

This reverberation device aims at both: Tight small rooms and ambiences well suited to modern drum and vocal productions up to large "epic" halls as known from high quality outboard gear. This reverbs sound ranges from rather concrete or even edgy up to smooth, transparent and artifact free reverb tails. It is designed for maximum flexibility and usability and to take place as a true high quality stereo main reverb.

epicVerb features two different sounding reverberation modes and 6 different stereo early reflection models. There are some standard reverb controls like 'TIME', 'DAMP' or 'PRE-DELAY' as well as reverb tail modulation and detailed control over very first reflections as an option.

It offers different reverb time handling for high and low frequencies as well as a musical sounding EQ section containing two "BootEQ" equalizers and additional high- and lowpass filtering.

Plug-in specification

- PC / VST compatible
- SSE and Assembler optimized sound engine
- State-of-the-art digital signal processing
- Smooth reverb tail processing without any ringing or metallic sounding artifacts
- Different reverb and early reflection modes for maximum flexibility

Quick Reference

PRE-DELAY	Pre-delay time in ms
TIME	Reverb decay time in ms
+10s	This switch adds 10 seconds to the 'TIME' parameter. This way one can obtain reverberation decay times from 10 up o 20 seconds
SELECT ER	Steps through the 6 different early reflection models (see below for further details). Selecting a specific ER also changes internally the size of the room model
DAMP	Damps the reverb tail (this actually affects reverb decay time)
REVERB/ AMBIENCE	EV features one 'REVERB' algorithm and in addition an extra 'AMBIENCE' mode. The ambience mode features just the ERs and no reverb tail
MOD	Dials in the reverb tail modulation (sometimes refered to as "chorus")
RT-LOW	The device offers different reverb time handling for high and low frequencies and this is done through RT-LOW which defines the reverb time for the low frequency range as a multiplier to TIME

RT-XOVR	Defines the crossover frequency according to RT-LOW
1st 2nd ER	Control about the very first early reflections is given with this parameter section: Changes timing and level of the early reflections behavior - use this to obtain a more focused or diffuse sound. It can change the overall sound as well ranging from a more 'colorful' to a rather 'transparent' sound. Changing the first affects the second and changing both affects other reverb details under the hood as well. This way altering the timing can affect the overall reverberation even if both 'LEVEL' controls are set to minimum (left most position). Timing is displayed in ms. Level control is unity in upper middle position and increases clock-wise (decreases counter clock-wise).
GAIN-FREQ, HiQ	The two EQ knobs: The outer ring selects the frequency and the inner knob increases or decreases that frequencies gain (+/-12dB or +/-18dB in HiQ mode). Inner knob in upper middle position is 0dB
HP-LP	Highpass and lowpass filter
MID-SIDE	Alters the width of the output signal by M/S processing
DRY-WET	Mixes the unprocessed (dry) and the processed (wet) signal
OUT	Output gain in dB
ON	Power on

The ER and Room Modes

HALL
- Large room size
- Widely distributed reflections
- Works best for larger reverbs

PLATE
- Medium room size
- Centered / even ER's
- More centered sound and reflection distribution

ROOM 1
- Smallest room size
- Mostly centered / even sound
- Best for smaller things and small plate simulations as well

ROOM 2
- Small room size
- Different and uneven ER distribution
- Works good as well for large halls

REFLEX
- Medium room size
- Offers more pronounced reflections
- Good for FX types of sounds or pseudo 'spring reverb'

ECHO FX
- Large room size
- Features some ghostly delays
- Cool for weird FX things

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